5 Way Active Audio Crossover With Integrated Equaliser

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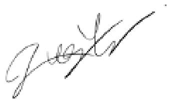
Tuong Bao Nguyen	
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1. Abstract

This project report presents the design and development of an audio crossover network, with an integrated five-band equaliser, and multiple speaker outputs. After conceptualisation, the design of the system uses a crossover with active filters to divide the input audio signal into five predetermined frequency bands (woofer, low-mid, mid, high-mid, tweeter), each directed through a dedicated equaliser and output stage. This report details the methodology, rationale, capabilities, limitations, trade offs, and areas of improvement for each module such as signal integrity, power handling, and component constraints, which is explored and factored into the initial design considerations. After finalising the design, the system is simulated and demonstrates effective frequency separation, impedance matching, controlled gain tailored to the speaker's power ratings, and volume control of specific frequency bands, capable of being adjusted to the user's liking. The design is then prototyped and vigorously tested under multiple conditions to ensure reliability and repeatability, and iteratively refined based on performance analysis, frequency response, power requirements, user feedback, and troubleshooting of observed discrepancies. All discrepancies, challenges, areas of improvements, and so forth are then discussed and justified in the context of the engineering design paradigm. A final version of the system was then constructed and then presented to the client with a live demonstration to showcase the system's functionality. Overall, results suggest a functional design that meets client's expectations, and exhibits stable and reliable performance.

2. Introduction and Conceptualisation

2.1 Context and Rationale

The goal of this project is to design, prototype, and test a five-way audio crossover that divides an audio input into low (woofer), low-medium, medium, high-medium, and high (tweeter), with an integrated multi-band equaliser, allowing for sound customisation by controlling the amplitudes of specific frequency bands. This project aims to improve users' audio experience by making sure each speaker operates within its optimal frequency range and allowing for fine tuning and customisation capabilities at different selected frequencies through the equaliser. The overarching design question to answer can be summarised as *"how can a five-way audio crossover with an integrated multi-band equaliser be designed, simulated, constructed, and tested using a systematic engineering design methodology while applying effective communication, teamwork and project management skills?"*

2.2 Project Planning, Review, and Research

The project heavily involves the integration of four key fields of knowledge: audio signal processing, filter design (passive and active), operational amplifier circuits, and engineering design procedures.

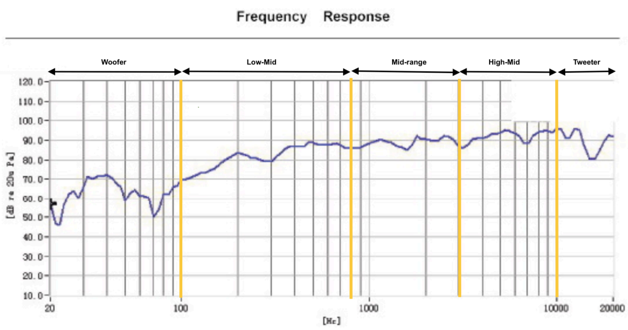


Figure 1: Frequency Response Graph of Enclosed Speaker

With respect to the current literature, there exists a number of different methodologies which may be used to determine the required values and number of frequency filters, as well as the frequencies and bandwidths for the equaliser. After conducting research in relation to audio crossovers, and comparing to common designs, the frequency ranges for the bass, low-mids, mids, high-mids and treble were chosen to be 0-100Hz, 100- 800Hz, 800-3000 Hz, 3000-10,000Hz and 10,000Hz+ respectively[1]. These frequency bands are chosen to not only be strategically spaced to cover the human audible spectrum (0 - 20,000Hz)[2], ensuring comprehensive audio coverage for any listener, but are also chosen based on the frequency response graph of the enclosed speakers (CE08483) used in the design as shown in Figure 1, allowing them to play within their appropriate frequency ranges[3]. This was also confirmed in a physical test using the AD2 and an audio sweep of 20Hz to 20kHz. The bass frequencies are connected to the woofer speakers, the mid frequencies to the mid-range speakers, and the treble frequencies to the tweeter speakers, acting as low-pass, mid-pass, and high-pass filters respectively.

For the integrated equaliser, due to interest and compatibility with the chosen design, the team has elected to implement the desired stretch feature of designing a 5-band audio equaliser, allowing for much finer control over specific frequency ranges[4]. After reviewing notable literature present within the field, the frequency bands are chosen to be the same as the crossover frequencies, strategically spaced to not just cover the human audible spectrum, but make a noticeable difference when tuning each specific frequency to the user's desire[5]. Active bandpass filters with op-amps will be used for each band[6]; the reasoning and design behind this will be covered in section 4 Design Overview and Evidence, of this report. Effort and time will be invested in this singular stretch goal to ensure that it works exceptionally well and is very customisable.

As for the actual circuit implementation, it is worth noting that for the project and associated presentation, audio will be played using an ASUS Zenbook Ryzen 7, where the manufacturer states that its audio jack has a typical output signal voltage of 1Vrms (max), 20-50mW, which will be heavily considered as part of the project design[7]. This could also be easily replaced with a wave generator for testing purposes as well.

With the overarching goal of creating a functional audio crossover with equaliser, the different necessary components of the project have been identified and divided into 3 main modules, broken down into their submodules, following the methodology covered in section 3 Methodology and Design Procedure, of this report. The specific details and discussions in relation to these modules, and engineering design procedure as a whole, are covered in section 4 Design Overview and Evidence, of this report, along with discussions relating to design tradeoffs, capabilities, limitations, and future areas of implementation.

Overall, this approach combines theoretical knowledge with practical application, mimicking real world testing and optimisation of audio performance.

3. Methodology and Design Procedure

The team approached this project using the project development cycle outlined in the *Engineering Design Workflow* in Figure 2[8]. This process conducted by the team was performed in 6 primary steps: 1. Conceptualisation, Project Planning and Research, 2. the Design Phase, 3. Simulation Phase, 4. Prototyping, Testing and Calibration, 5. Analysis and Iteration, and 6. Finalisation and Presentation phase. All six steps are detailed and delivered within this project report.

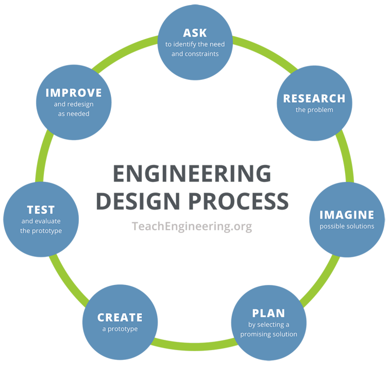


Figure 2: Engineering Design Workflow

Through a self-conducted workshop session, the team followed this methodology in the conceptualisation, project planning, and research phase:

1. Review the 'ENAD Design Project' files (*ENAD Design Project Description, ENAD Design Project - Assessment, etc.*) to understand and attain context for what is being designed and the client's expectations of the project.
2. Research audio networks and their associated power management circuits individually, and discuss them as a team.
3. Research the key principles of audio crossover design and audio equalisation with respect to acoustic properties and the typical frequency ranges of speakers as a team.
4. Determine, as a team, the stretch goal, the crossover frequency points and equaliser frequency bands necessary.
5. Delegate tasks for the following project phases, appropriately and effectively.
6. Plan project timelines and milestones, as a team, to ensure everything is completed to a high standard within timeframe, and meets client expectations, with the use of progress tracking methods like weekly meetings/workshops and updated gantt charts.

The Design Phase, which followed the project planning, research, and conceptualisation phase was conducted firstly by individuals following the tasks delegated in the planning phase, then shared and discussed their design work via online communication and in a follow up meeting. The methodology followed by the team in this phase is outlined below:

1. Brainstorm and compare different possible design ideas, comparing capabilities, trade offs, and limitations of the designs which are explored below.
2. Determine one singular design out of those proposed, agreed by the entire team, that precisely solves the engineering design problem, in the context of audio crossover networks and equalisers.
3. Determine the different modules and submodules that are required for the selected design.
4. Design each module individually, explaining the purpose and rationale of the module and its design process.
5. Selecting components and their values that are required for each individual module, factoring in cost, performance, repeatability, accuracy, etc.
6. Combine the modules, and create a block diagram representative of the entire design.

The simulation phase, which followed the design phase, was conducted as a team within a self-directed workshop session, and follows the methodology below:

1. Using the proposed design from the previous phase, the system's response to different frequency ranges as well as its inputs was modeled using LTSpice, first as individual modules to ensure all modules work as expected, then as a combined audio crossover and equaliser.
2. Crossover points and equaliser settings were tested via simulation on LTSpice.
3. An additional desired feature the team agreed to implement was also tested to ensure it was theoretically sound and functional.
4. The system was validated through simulation and the results of the simulation for both the individual modules as well as combined audio crossover and equaliser system were discussed for discrepancies and their justifications.

The prototyping, testing and calibration phase, which follows the simulation phase, was conducted individually with the respective module in another self-directed workshop session, then combined as a team to attain a fully functional prototype of the entire audio crossover and equaliser system. This was done using the following methodology:

1. Using the same proposed design from the design phase, each module was built independently on a breadboard, along with the chosen desired feature, in order to ensure that it works as expected, and to easily troubleshoot any issues that arose, and discuss initial findings with each module.
2. Combine all the modules to form a fully functional audio crossover and equaliser system, allowing for real time audio adjustments, and discuss their interactions when connected to the speakers and input audio sources.
3. As a team, fine tune the crossover and equaliser settings for optimal sound quality, while ensuring that connections are secure and that the circuit is robust.

The analysis and iteration phase, which follows the prototyping, testing and calibration phase, was conducted as a team in the same workshop session, using the prototype that was constructed in the previous phase, and follows the methodology below:

1. Using the AD2, each individual module of the circuit was analysed independently, to measure the frequency response of the respective module.
2. This was compared to the audio output with and without the crossover network and equaliser to demonstrate the effect of each of the modules.
3. Using the AD2, the entire audio crossover and equaliser system was tested as a whole to measure the frequency response of the overall system from audio input to speaker output.

4. Comprehensive and detailed measurements of frequency response, audio range, and signal integrity were recorded for both individual modules and the entire system using a number of tests, including the Waveforms network analyser and spectrum analyser, to ensure reliability and repeatability of results.
5. These results were then discussed with discrepancies being justified for each individual module as well as combined audio crossover and equaliser system.
6. Iterative design adjustments based on frequency response, audio performance, power requirements, and user feedback were conducted.

The finalisation and presentation phase, which follows the analysis and iteration phase, was completed individually and then discussed in an online meeting via online communication methods. The methodology followed by the team in this phase is outlined below:

1. A final version of the circuit was constructed with consideration for cost, performance, robustness, durability, aesthetics, as well as user friendliness.
2. A live demonstration that allows for clear and audible differentiation between frequency bands was then prepared.
3. The oral presentation was prepared with appropriate media tools along with practicing communication of the project's features and results.
4. The team project report was finalised and made deliverable, ensuring comprehensive documentation of methodology, iterations, findings, analysis, and discussion.

4. Design Overview and Evidence

4.1 General Design Discussion

Following the conceptualisation phase, several design ideas were proposed by the team, each with different advantages, capabilities, trade offs, technical challenges and limitations that sought to answer the design problem. Hence, critical evaluation and balance between simplicity, performance, and flexibility was vital for choosing a certain design. During this phase, key factors that influenced the final design involved availability of components, signal integrity, power consumption, circuit complexity and layout, repeatability, and compatibility with the audio device and speakers as well as other modules were carefully considered. Through iterative discussions, the team determined the feasibility of each of the ideas, narrowing down to the following solution that best addressed the client's performance and functionality requirements.

The final design is composed of three main modules, each designed independently first to ensure ease of troubleshooting and flexibility for future modifications. The way that these modules fit together is that when combined, the audio is inputted using an aux cable from a laptop to an aux jack at the beginning of the input stage. The input stage blocks DC and prevents loading effects on the signal source, feeding into an audio crossover module to divide the frequencies five ways. This then feeds into a 5-band equaliser module to allow for volume control over a range of frequencies and handles the audio in relation to impedance and power handling for the speakers, and the output connects each audio signal to their respective speakers. The block diagram can be seen in Figure 6. The specifications and rationales of each module is discussed below. For this design, the audio op amp LM4562 has been chosen for its high performance, low distortion (0.00003%) and low noise (2.7 nV/√Hz) which is good when dealing with audio signals and with a high slew rate $(20V/\mu s)$, it is responsive for audio volume changing applications[9]. A notable drawback with using this op amp is its output swing headroom since it cannot fully swing to the supply rails (generally 1.5V from each rail)[9]. However, with $V_{CC} = +\ 5V$ and $V_{EE} = -\ 5V$, and given the laptop audio jack typically provides a max 1Vrms, which is higher than standard for most laptops, the op amp provides sufficient headroom despite the notable limitation.

Overall while this design meets the client's expectations and core objectives of the project, with capabilities in adjusting the overall volume levels as well as volume levels of predetermined ranges in the audio spectrum, and dividing the audio into bass, midranges and treble, there are inherent limitations with this design. These further capabilities, trade offs and limitations will be explored in much greater depth below along with areas for improvement, rationales, and initial design considerations.

4.2 Module Design and Discussion

Module 1: Audio Input Stage

This module serves to connect the audio signal to the following modules of the design. It is composed of four main components including the aux jack for inputting audio, DC blocking, a master volume control, and loading prevention. The corresponding block diagram can be seen in Figure 6. This design was chosen due to its capability in maintaining signal quality, removing unwanted noise and DC offset, preventing signal loss, providing impedance matching, and full volume control. However, limitations and key tradeoffs for this module include possible signal attenuation from the DC blocker, especially at lower frequencies. This is a result of the DC blocking capacitor having a high pass filter behaviour. However this tradeoff is necessary in order to prevent possible signal shifting. The potentiometer necessary for the master volume control can potentially introduce noise, and may wear over time, causing impedance mismatches that affect the signal integrity. A further limitation with this design is that since there is only a buffer amp with gain of 1, the volume can only be decreased rather than both decreased and increased due to a lack of a preamplifier. However, the equaliser aims to address this, and initial simulations and physical testing with different tracks shows that it is capable of performing well despite these limitations.

Initial designs included a pulldown resistor after the potentiometer in case the wiper within the potentiometer was unable to completely ground the signal which meant it would not be able to fully mute the audio. However, this detail was not implemented since after physically testing the chosen potentiometer, it proved to reliably ground the signal without the need of a pull down resistor, and when using this pull down resistor in simulation, it would slightly degrade the signal if the resistance value was not perfect, introducing unwanted distortions. Too high of a value relative to the potentiometer would virtually do nothing, and too small of a value would significantly alter the signal. Physical resistors on average have a variance of $\pm\ 5\ -\ 10\%$ and for large values of resistors which is necessary, this could add or reduce $10k\Omega$ to a $100k\Omega$ resistor for example[10]. The variation is too great for the value that is needed. This may be explored again when prototyping if necessary.

For future improvements and implementations, signal mixing will be explored using multiple audio devices and summing amplifiers, but at the current moment, a lack of aux jacks prevents this implementation, along with better components more suited for the task like a higher quality aux jack to reduce the noise and signal loss. The current Aux jack (FC68131) has poor contact pressure which may affect the input signal, and has a limited current rating (10-15A)[11]. Additionally, a visual representation of the volume control like an LED strip or seven segment display was also considered but due to its complexity with using seven different comparators and a 7 segment decoder IC like the 74LS47, this was not implemented as it not only added additional cost and complexity, but could also affect the loading and signal integrity with noise introductions[12]. Also, a pre amplifier was considered but was deemed unnecessary due to the already high laptop output voltage.

Description	**Component Specification**	**Purpose and Rationale**	**Equations and Evidence**
Aux jack	FC68131	The aux jack provides an analogue audio input for connecting the laptop to the rest of the modules. It is simple, compact, and cheap which makes it good for this purpose[11].	N/A
DC blocking	$C1\ =\ 10\mu F$	The DC blocker is a capacitor which removes DC offset from an audio signal, acting as a high pass filter to block DC and allow AC signals to pass. This is important since it prevents signal shifting or saturation which leads to unexpected outputs[13]. The capacitor value was chosen so that it minimises the cutoff frequency (f_c), with respect to the high impedance of the buffer amp, as to not alter the input signal to a significant degree.	$f_c\ =\frac{1}{2\pi RC}$ The LM4562 has very high input impedance, tpycially $300k\Omega$, so having $C1\ =\ 10\mu F$ gives f_c=0.053Hz which will not significantly affect signal.
Master Volume Control	$10k\Omega$ potentiometer	The potentiometer allows for control of the volume across the entire audio. The volume can be reduced or even muted. The component value of $10k\Omega$ was chosen since it balances being high enough to avoid heavy loading and low enough to avoid adding too much noise to the signal[14].	$H(s)\ =\ \frac{sRC}{1+sRC}$
Loading Prevention	Buffer amp (1x LM4562)	The buffer amplifier has very high input impedance and very low output impedance, preventing loading effects between the five filters of the next module. This is necessary for loading prevention, preventing signal distortion and loss, and reducing interference with the following modules, and other unexpected behaviours[15].	$H(s)\ =\ 1$

Module 2: 5-Way Active Audio Crossover Filters

The 5-way crossover module was designed to divide a full-range audio signal into five distinct frequency bands: bass, low-midrange, mid-range, high-midrange and treble. This type of crossover was taken into consideration with the additional desired feature of implementing a 5-band audio equaliser, and is capable of allowing for each frequency band to be independently processed and amplified. The goal of this crossover was to achieve smooth transitions between frequency bands, with a steep rolloff rate after the crossover point to properly separate the frequency bands. Figure 6 shows the block diagram for the 5-Way Active Audio Crossover.

The initial approach involved using RLC circuits for passive filtering due to its simplicity, however, factors such as limited control over frequency response, low gain and filtering characteristics being affected by source impedance led the team to abandon the approach early in the design phase[16]. Moreover, due to the bulk and high cost of inductors at low frequencies, this design opts for practicality offered by capacitors. After exploring active filters, the Sallen-Key topology, which has the advantage of requiring fewer components and supporting unity gain operation, which minimises signal distortion, was adopted[17]. Each filter stage comprises a pair of resistors and capacitors arranged around an op-amp, forming either a low-pass or a high-pass second-order filter, which can be cascaded together to form a second-order bandpass with a 12dB/octave roll off rate[17]. Cascading two identical Sallen-Key filters results in a fourth-order response, which has a steep 24 dB/octave roll off rate[18]. This has been implemented in the woofer and tweeter to isolate these frequencies more, as they are more susceptible to distortion and people tend to like bass-boosting their songs. Second-order filters for the band-passes were implemented on the midrange bands as it had sufficient roll-off to isolate frequency bands, without increasing the power demand, complexity, and cost of the circuit (as implementing a fourth-order bandpass would comprise of 4 Sallen-Key topologies cascaded together). Considering the division of the audio signal into five frequency bands, it was essential to implement a filter with sufficiently steep attenuation slope to isolate each band by preventing unwanted frequencies from spilling to adjacent speaker drivers. Lower-order filters, such as first-order (6 dB/octave) designs were deemed unsuitable in this context due to their more gradual attenuation, which would allow for more frequency overlap between adjacent bands[18]. This issue becomes especially problematic in a 5-way crossover system, where the available bandwidth per band is narrower and the risk of interference between frequency bands is significantly higher. Higher-order filters, on the other hand, did not show a great enough increase in performance to justify the additional cost as well as power requirements. This was tested theoretically using LTSpice and their frequency response was obtained. Hence, the selected design strikes a good balance between performance and cost.

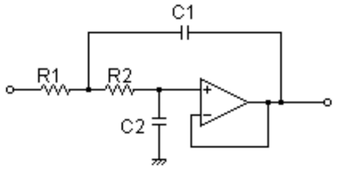


Figure 3: The schematic for a unity-gain low-pass Sallen-Key filter.

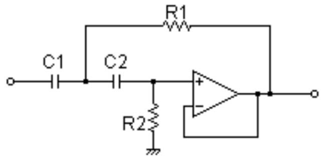


Figure 4: The schematic for a unity-gain high-pass Sallen-Key filter.

To calculate the cutoff frequencies for each band, the formula for a second order Sallen-Key configuration was used[19]. The schematics of these configurations are shown in Figure 3 and 4. Assuming equal resistor and capacitor values ($R_1 = R_2$ and $C_1 = C_2$), the equation simplifies to:

$$f_c = \frac{1}{2\pi\sqrt{R_1 R_2 C_1 C_2}} = \frac{1}{2\pi RC} \quad (1)$$

For each filter, component values were selected accordingly to match the frequency range of each band: bass (<100 Hz), low-midrange (100-800 Hz), midrange (800 Hz - 3000 Hz), high-midrange (3000-10,000 Hz) and treble (>10,000 Hz). The reasoning behind choosing these cut-off frequencies were previously explained in section [2.2 Project Planning, Review, and Research] of this report. The quality factor, Q, indicating the sharpness and selectivity of the filter was also considered and calculated with the equation[17]: $Q = \frac{1}{3 - A}$, where A is the passband gain. A unity-gain configuration (A = 1, Q = 0.5) was implemented for the woofer and tweeter which gives smoother transitions between bands. A Butterworth configuration (A= 1.586, Q = 0.707) that achieves a maximally-flat passband was implemented for the three bandpasses which allow for improved stability and lower phase noise.

Therefore, for the unity-gain configuration, the voltage gain of the filter circuit is 0.5, or -6 dB (critically damped) at the cut-off frequency point[18]. While for the Butterworth configuration, the voltage gain of the filter circuit is 0.707, or -3dB (underdamped) at the cut-off frequency point. The filter stages are set for unity-gain, so the overall gain of the crossover is 1, as the goal was to maintain the original signal level without any distortion. To achieve unity gain in the bandpass, a voltage divider was used by adding a resistor.

While the module is capable of crossover applications with fixed frequency bands, it does have limitations. The crossover frequencies are determined by fixed RC values, meaning any changes in the desired cutoff frequencies would require changing the physical components. This lack of flexibility makes it difficult to adapt the module for different speaker configurations. Additionally, the reliance on multiple op-amps throughout the circuit introduces issues such as thermal drift, added noise, higher power consumption and variations in op-amp characteristics due to temperature fluctuations, which may be an issue with having so many filters close together[20]. In future revisions, this design could be enhanced by incorporating potentiometers to allow for variable crossover frequencies without hardware changes.

Description	**Component Specification**	**Purpose and Rationale[17]**	**Equations and Evidence[17]**
Low-pass Filters	2x LM4562 4x $15.92k\Omega$ resistors 4x 100nF capacitors	This filters out high frequencies with a cutoff of 100Hz. Given the desired cutoff frequency of f_c = 100 Hz, capacitor value, C = 100nF was chosen first from the kit, then the equation (1) is rearranged to solve for the resistors value R, $R = \frac{1}{2\pi C f_c} = \frac{1}{2\pi \times 100 \times 10^{-9} \times 100} = 15.92k\Omega$	$H(s) = \frac{K\omega_0^2}{s^2 + \frac{\omega_0}{Q}s + \omega_0^2}$ Refer below.
3 x Bandpass Filters	6x LM4562 Low-mid: Refer to the right for capacitor and resistor values.	Each bandpass filter is designed by cascading a high-pass and low-pass Sallen-Key filter stage to form a bandpass. Given the desired low and high cutoff frequencies f_L and f_H , the capacitor value of each bandpass was chosen first, and the corresponding resistor value is found using equation (1): • Low-midrange: f_L = 800 Hz, C_L = 100nF ⇒R_L= 1.99kΩ f_H = 100 Hz, C_H = 100nF ⇒R_H= 15.92kΩ • Midrange: $f_L = 3k\ Hz$, $C_L = 10nF \Rightarrow R_L = 5.3k\Omega$ f_H = 800 Hz, C_H = 100nF ⇒ R_H= 1.99kΩ • High-midrange: $f_L = 10k\ Hz$, $C_L = 1nF \Rightarrow R_L = 15.92k\Omega$ f_H = 3k Hz, C_H = 10nF ⇒ R_H= 5.33kΩ	$H(s) = \frac{Ks^2}{s^2 + \frac{\omega_0}{Q}s + \omega_0^2}$ Where: Q is the quality factor, K is the gain factor and ω_0 is the characteristic frequency (rad/s)
High-pass Filters	2x LM4562 4x $15.92k\Omega$ resistors 4x 1nF capacitors	This filters out low frequencies with a cutoff of 10kHz. Given the desired cutoff frequency of f_c = 10,000 Hz and selecting the capacitor value C = 1nF, $R = \frac{1}{2\pi C f_c} = \frac{1}{2\pi \times 1 \times 10^{-9} \times 10000} = 15.92k\Omega$	$H(s) = \frac{Ks^2}{s^2 + \frac{\omega_0}{Q}s + \omega_0^2}$ Refer above.

Module 3: 5-Band Audio Equaliser and Output Stages

This module directs the five outputs of the crossover module into their respective speakers, with equalisers and volume control, seen in the block diagram, Figure 6. Each crossover output feeds into an active equaliser circuit, where each band uses one LM386 power amplifier, chosen for its small component count, specialisation in audio applications and adjustable gain of 20-200.[21] This high gain is critical since the voltage output of the crossovers may be low due to the significant number of op-amps, low input audio signal, and parasitic effects of passive components used in the design[22]. A drawback with having 8 pins is the more complex physical implementation, but this tradeoff is deemed necessary for the signal-boosting capabilities[21]. The input signals flow through a buffer amp which prevents loading effects between the stages, where the crossover output might drop in voltage when connected directly to the equaliser input, along with keeping the signal undistorted[15]. It then goes to the potentiometer

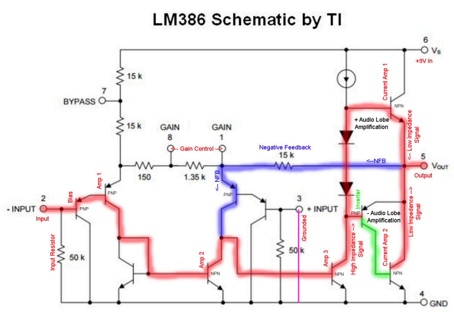


Figure 5: The schematic for the internal components of LM386 Power Amplifier

for volume control, and is grounded through the non inverting input to stabilise DC levels[14]. Next, there is a pull-down resistor for current limiting and a DC-coupling capacitor to block DC offset. The LM386 amplifies the signal through pins 3 and 8, and the output of the op-amp then flows through a large electrolytic capacitor to prevent any DC signals flowing into the speaker[13]. There are also two power supply bypass capacitors across Vcc and ground to filter out additional noise and DC offsets which has a limitation of slightly attenuating low frequencies due to high pass-like behaviour. However, this tradeoff is deemed necessary to remove unwanted noise which is crucial for audio applications.

To help ensure the sound quality and signal stability, sufficient headroom was left, minimising any distortion. Another major consideration was the impedance for the speakers. Having amplifier impedances that don't match the speakers' may cause errors and distortion. Hence, initial designs used a Zobel network connected in parallel to the speaker[24]. This network seemed to cause a few issues during the simulations due to distorting the signal so may be removed. The resistor in this network is chosen to be slightly greater than the speaker impedance (10Ω vs 8Ω). This stabilises the amplifier as at high frequencies, the increasing impedance of the speaker due to inductance shorts the resistor in the Zobel network, maintaining the stability of the amplifier[25]. Another major consideration was making sure that the amplifier power did not exceed that of the max rated power of the speakers since improper power handling may cause an unexpected output or even speaker damage. For the AD2 this is not a major concern since it is only able to supply a maximum of 5V[26], but the lab equipment can supply up to 30V[27]. To combat this, the total gain of the circuit was calculated to ensure that it was less than the power rating for the speaker, which is 5Wrms, 8Ω.[3]

$$P = \frac{v^2}{R} \Rightarrow v = \sqrt{PR} = \sqrt{5 \times 8} = 6.32Vrms$$

Hence, considering the max audio input from the audio jack of 1Vrms, the overall gain of the circuit is controlled to be less than 5 (5Vrms), considering headroom. A notable limitation is that the output does not reach the maximum possible volume, due to the impedance of the circuit elements and preventing the speaker from possibly going above its power rating, which will be considered for the prototyping phase. Some further limitations and factors that were considered are the component tolerances (resistors, capacitors), inherent noise of the LM386 and power dissipation of the LM386[21]. While they do have some impact on the signal strength and clarity, they are difficult to guard against and are minor in severity for the simulations. Hence, while they are noted, there are no countermeasures present for the time being unless they become greater issues in the implementation, prototyping and testing phase.

Initial designs considered using summing amplifiers to combine the low/low-mid bands, and the high/high-mid bands as the highest and lowest frequency bands tend to be less common for most audio signals. However, it was decided that it is more logical to use five speakers due to flexibility, practicality, edge case considerations (some audio signals have a lot of very high or low frequencies) and ease of troubleshooting. During prototyping, this idea may be revised if necessary. A major tradeoff with this design is its dependency on the crossover frequencies for simplification. However, for future implementations, a parametric equaliser may be explored to work around this tradeoff in dependency.

Description	Component Specification	Purpose and Rationale	Equations and Evidence
Buffer Amp	LM4562	The buffer amplifier has very high input impedance and very low output impedance, preventing loading effects between the modules. This is necessary for loading prevention, preventing signal distortion and loss, and reducing interference before getting amplified by the equaliser, along with preventing other unexpected behaviours[15].	$H(s) = 1$
Volume Control	5 x $10k\Omega$ potentiometers	The volume control allows users to adjust the gain to boost/cut the audio signal at different frequency bands by turning the dial on each potentiometer to change the volume. This is implemented via five potentiometers, each connected to one of the crossover outputs. The potentiometers are chosen to be $10k\Omega$ in order to minimise noise (compared to higher valued potentiometers) and to ensure reduced loading between other modules. They also offer a wider range of control for the audio signal in comparison to lower valued potentiometers while being readily available, being the standard value for audio applications[14].	$H(s) = \frac{sRC}{1+sRC}$
Equaliser	5 x LM386 power amps 5 x 1000µF capacitors 10 x 100µF capacitors 5 x 100nF capacitors 5 x $10k\Omega$ resistors	The five audio signals from the crossover are each put through an LM386 power amplifier to boost the gain to the default value of 20. The gain control is between pins 1 and 8. When nothing is connected externally, the gain defaults to the value of 20 but is limited to Vcc (5V), as the current must flow entirely through the 1.35kΩ resistor, seen in Figure 5[21]. The 1000µF output capacitor works with the 8Ω speaker to block DC and allow AC signals to pass[13], acting as a highpass filter. The capacitor value was chosen so that it minimises the cutoff frequency (f_c) to be as close to 20Hz (audible human spectrum) as possible, so the 1000µF capacitor is used, seen in equations and evidence. The two decoupling capacitors across Vcc and ground help filter out noise. The 100µF capacitor filters out low-frequency noise while the 100nF capacitor filters out high-frequency noise. The Zobel network has a resistor and capacitor in series to stabilise the amplifier at high frequencies[23]. The Zobel resistance (R_z) is chosen to be slightly higher than the speaker impedance (10Ω vs 8Ω respectively). Hence the Zobel capacitance (C_z) is chosen to be 0.1 µF, as seen in equations and evidence. The power supplied to the power amp is then set at 5V for appropriate powerhandling with the CE08483 speakers.	$f_c = \frac{1}{2\pi RC}$ $= \frac{1}{2\pi \times 8 \times 1000 \times 10^{-6}}$ $= 19.894\ Hz$ $C_z = \frac{1}{2\pi f R_z}$ $= \frac{1}{2\pi \times 20 \times 10^3 \times 10}$ $= 0.796\ \mu F$
Speaker Output	5 x CE08483 speakers	The speakers transform the audio signal from the equaliser into sound waves that can be heard by the human ear by vibrating the drum[3].	N/A

4.3 Audio Crossover and Equaliser Design Finalisation

After carefully designing each module separately, they can be combined together to make the overall audio crossover and equaliser design shown below. The three different modules have been outlined as well. The discussion of how each module specifically fits with one another was covered in section [4.1 *General Design Discussion*] of this report.

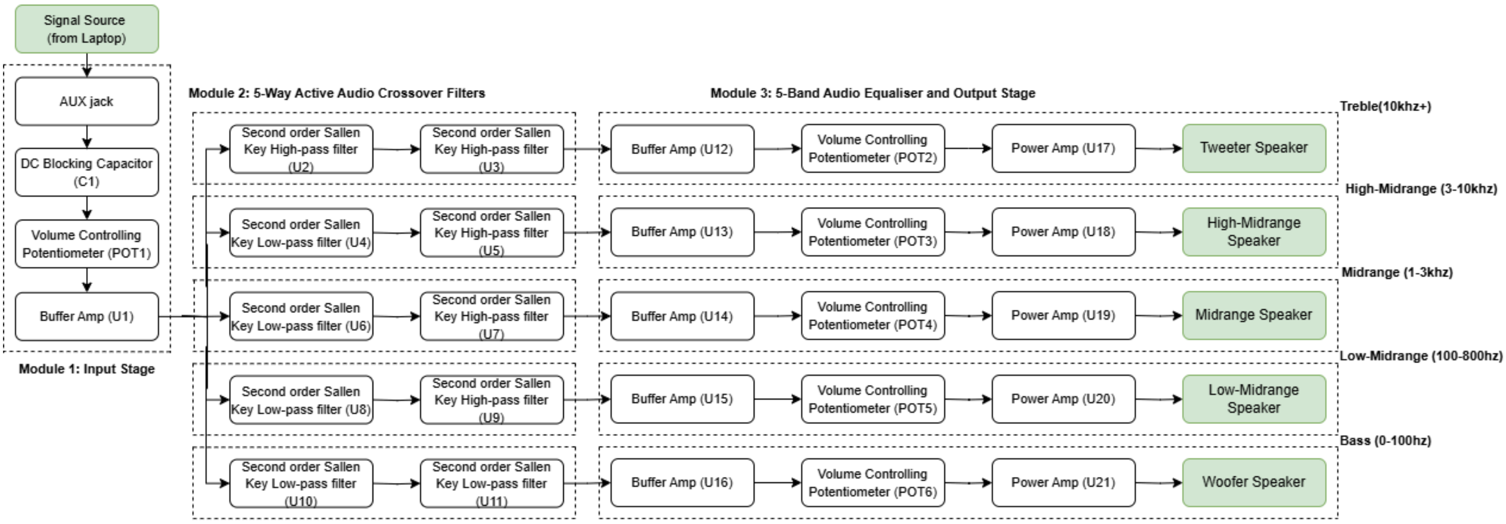


Figure 6: Initial Block Diagram of Audio Crossover and Equaliser

5. Development and Simulation

Each module is first developed into an appropriate schematic then simulated individually to ensure that the design performs as expected. Then a simulation for the entire audio crossover with equaliser circuit is conducted to ensure that the different modules are compatible with one another. Note that the input signal has been chosen to be a [wav file] to match real world audio inputs, covering a range of frequencies (20Hz - 20kHz to cover the human audible spectrum) compressed down to 3 seconds (from 35 seconds).

5.1 Module Simulation

Module 1: Audio Input Stage

The schematic, Figure 14, was created based on the design in the previous section. This provides the simulations seen in Figure 7. The audio sweeps from 20Hz to 20kHz which explains why the wavelengths start wide and gets shorter over the 3 second duration. With the red trace as the input and using the '.step param', to turn the potentiometer, it can be seen that as the potentiometer resistance increases, the magnitude and voltage decreases proportionally and therefore so does the sound level, while the signal maintains the same shape. This agrees with the theory. Translating this to the frequency domain in Figure 8, it can also be observed that increasing the resistance, by turning the potentiometer, lowers decibel as well, while maintaining the same input audio signal shape as desired. This demonstrates that the potentiometer is capable of being a master volume control. When the potentiometer is fully turned (WIPER=1), the signal is grounded so the audio is completely muted as expected, represented by the pink line at 0mV in the time domain, and being virtually invisible in the frequency domain. The proportion that it decreases aligns with the transfer function in the theory, $H(s) = \frac{sRC}{1+sRC}$, but not exactly, which may be a result of limitations present within the files used which will be explored in section [5.3 *General Design Discussion*]. This also further confirms the theory that the volume is only able to be reduced, not increased as previously discussed during the design of the module, with the wiper set to both 1 (no resistance, pink trace) and 0 (max 10kΩ resistance, red trace). The capacitor was also capable of blocking the DC voltage after simulating a test 500mV DC offset from the voltage source, also confirming the theory behind the DC blocking capacitor as well.

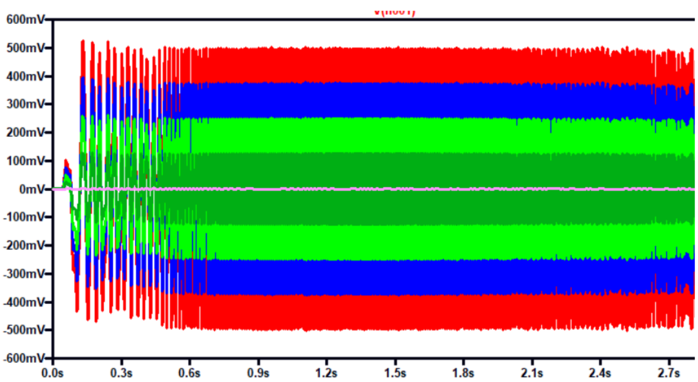


Figure 7: Time Domain Simulation of Input Stage

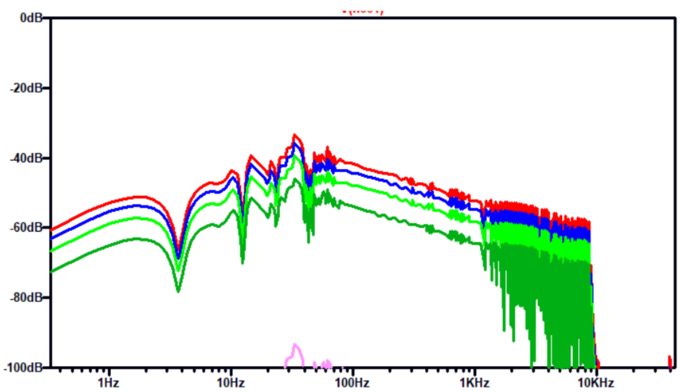


Figure 8: Frequency Response of Input Stage

Module 2: 5-Way Active Audio Crossover Filters

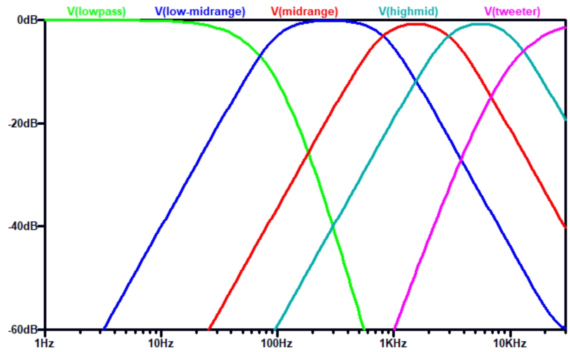


Figure 9: Bode Plot of Theoretical 5-Way Audio Crossover

Figure 14 shows the complete schematic of the 5-way audio crossover, where each filter stage handles a specific portion of the audio spectrum (each stage has been labelled). The simulation results are shown in Figure 10, which illustrates the behaviour of the crossover when a 3 second frequency sweep from 20Hz to 20kHz is applied (represented by the input grey trace). As expected, the output from each filter stage rises in amplitude only when the input frequency enters that filter's passband. For example, the bass output (green trace) starts off with a strong amplitude (~400mV) in the beginning of the sweep when the input signal is around 20 Hz to 100 Hz. As the frequency increases beyond the bass range, the output gradually attenuates, which aligns with the theoretical roll-off of 24 dB/octave for the woofer and tweeter, and 12db/octave for the midranges, seen in the Bode plot of Figure 9. This simulation agrees with the theory discussed in section [*4.2 Module Design and Discussion*] and provides a visual representation of the filter's behaviour depending on the frequency. The low-midrange output starts rising at around 0.5 seconds as the bass declines and begins attenuating at around 0.7 seconds. The mid-range output takes over from around 1.2 seconds and continues into the middle of the simulation. The high-midrange and the treble output remain near zero at the start of the simulation but begin rising in amplitude at around 1.8 and 2.5 seconds respectively, as the input signal reaches high frequency bands in the 3-20kHz range that it is designed to pass. Overall, this simulation is indicative of a successful and capable crossover design as similar behaviour can be seen in the frequency domain in Figure 11 as well, where the appropriate band increases in magnitude when the input signal reaches its respective frequency ranges.

The simulation also shows the overlap between adjacent frequency bands, resulting in multiple outputs being active over the same segment of the sweep. This is a consequence of the Sallen Key filters where they begin gradually attenuating frequencies before and after their cutoff points[18]. This results in a transitional region where adjacent filters are passing the same frequencies, but at different attenuation levels, which is good to help prevent jarring changes in the frequency response, which is rather common in modern day pop music.

Whilst simulating, a limitation was noticed that destructive interference occurs, resulting in signal loss in some regions where the overlapping outputs do not add coherently, as seen in the simulated Bode plot in Figure 11. While resembling the shape of the theoretical, this issue stems from the phase differences between each filter stage, where signals become out of phase with each other where they overlap[28]. This results in a reduction in volume, particularly in the treble region. This can be addressed with the use of all-pass filters that adjust the phases so that there is at most 3dB variance, which is the standard for commercial speakers.

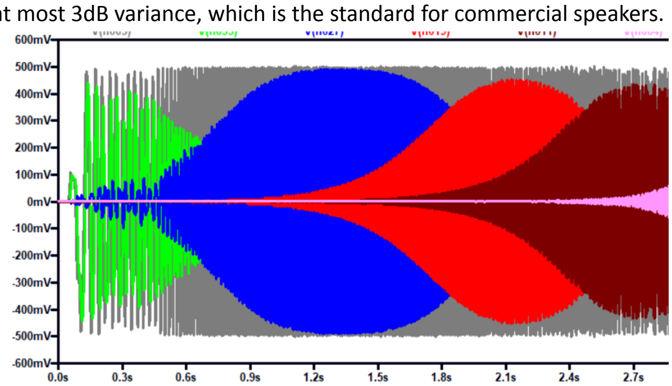


Figure 10: Time Domain Simulation of 5-Way Crossover

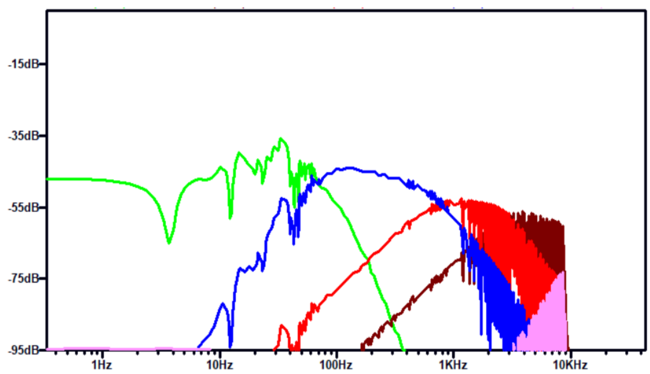


Figure 11: Frequency Response of 5-Way Audio Crossover

Module 3: 5-Band Audio Equaliser and Output Stage

Figure 14 shows the complete schematic of the equaliser and output stage (out of a total of five identical modules). After simulating, this provides the time-domain graph in Figure 12 which shows that the signal is successfully amplified by 4.90 on average (found using cursor averages at different peaks; $\frac{2.35V}{0.48V}$) which is below the max power rating of the speaker (6.32Vrms) as designed. This is not exactly the theoretical gain of 20 due to loading effects present when combined with the speaker. After removing the speaker and its impedance, a greater gain (closer to 20) was achieved but the signal saturated at about 4V due to the headroom of the amp. Additionally the LM386 file is not perfectly suited for LTSpice simulation which will be discussed in section [*5.3 General Design Discussion*] below. The audio file itself also contains a significant amount of noise which became exaggerated when the original 35 second audio file got compressed down to 3 seconds and then amplified. This may explain the small unexpected fluctuations. However, the plot shows the equaliser outputs acting linearly, indicating that there is no distortion due to the ideal nature of the simulation. The input sound file tested consisted of audio signals from low to high frequencies at roughly 1Vpp (green trace), and the equaliser outputs show that the speaker has considerably higher output voltages for the different frequencies, but never exceeding the max voltage that the speaker can handle, which is logical. Considering that the audio signal is the same for previous modules, it can be seen that the speakers handle the same frequency bands as the crossover section, but the amplitude of the signals are amplified, indicative of a successful amplifier design. A tradeoff is that the gain for lower frequencies appears to be higher due to higher voltage swings and current demands on the power amplifier for the lower frequencies. Low frequency waveforms like bass have longer cycles and carry more energy per cycle, and so more power is needed to move the speaker cones. However, the speaker output is noticeably amplified, which can be seen for both the time and frequency domains in Figure 12 and 13. It can be seen that as the potentiometer is turned and its resistance decreases, the volume is amplified according to its transfer function. It starts as the green trace (input signal), then amplifies to the blue trace, and when at max volume, it becomes the red trace. It is worth noting that at the beginning of the signal, the original input seems louder for less than 8Hz, as shown in Figure 13, but then gets overtaken as it is amplified. This makes sense since the use of capacitors at the outputs needs to be charged before allowing full signal swing, so the output starts weak in both time and frequency graphs.

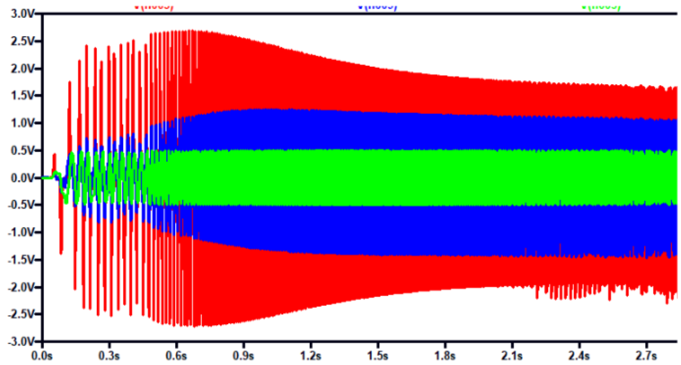


Figure 12: Time domain simulation of 5-Band Equaliser

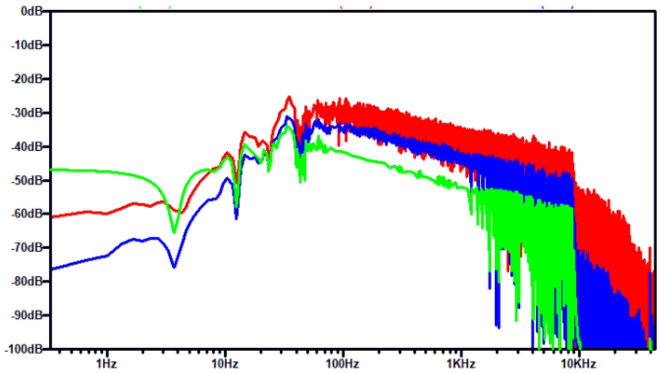


Figure 13: Frequency Response of 5-Band Equaliser

5.2 Combined Crossover and Equaliser Simulation

Now that each module has been simulated and shown to work, they can be combined into the following schematic of the combined audio crossover and equaliser in its entirety as shown in Figure 14. Using this schematic, the combined design of the crossover and equaliser may be simulated. Figure 15 shows an example of the functionality of the design. For this specific example, the master volume control has been set to the max. As for the equaliser settings, the bass has been set to the max. The low-mid has been completely muted. The mid is set to 0.8×max, and both the high-mid and treble frequencies are set to the max. The input audio signal has been included for comparison. This selection is chosen to test and demonstrate both the functionality of the crossover as well as the equaliser. When converted to the frequency domain, this same test can be seen in Figure 16.

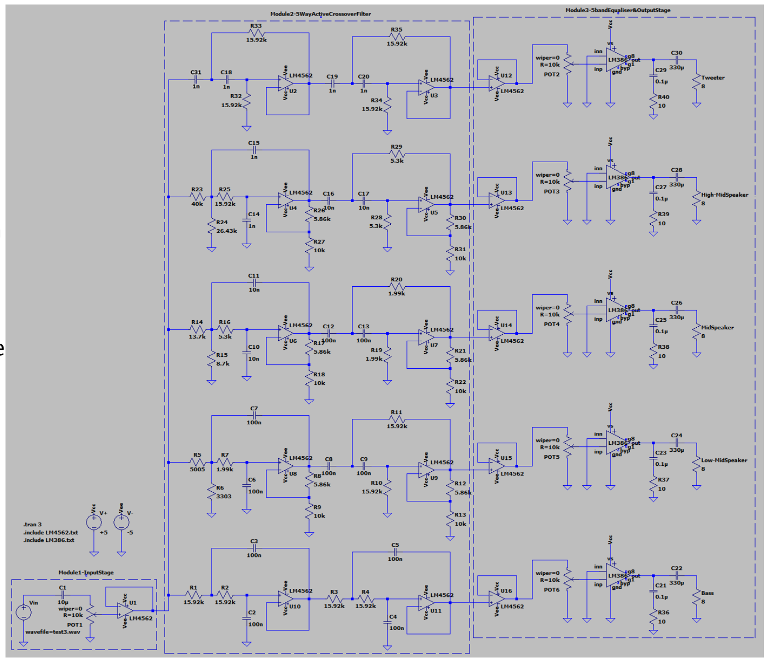


Figure 14: Completed Schematic Combining all Sub-modules (Enlarged ver. in Appendix A)

5.3 General Simulation Discussion

The goal of this design was to be able to control the volume of an audio input signal at selected frequency bands. This simulation of the entire design agrees with the theory and as a whole, demonstrates its capability in acting as an audio equaliser by allowing users to change the volume at specific frequencies. As can be seen in Figure 15, the crossover has successfully divided the signal where each coloured trace represents a range of frequencies. The input signal (grey) has also been plotted for reference. The equaliser is equally successful as for the test input signal, the bass (red, <100Hz) has been boosted to the max which can be seen in Figure 15 and 16. With an input voltage of about 1Vpp, it has been boosted to about 5Vpp. Now this result is not expected, but can be explained as a result of the lower frequencies requiring higher voltage swings and current demands on the power amplifier. Low frequency waveforms have longer cycles and carry more energy per cycle, and so more power is needed to move the speaker cones, which is a tradeoff of this chosen design. Now the low-mid (dark blue, 100-800Hz) has been completely muted. Hence, it does not appear in the frequency domain, and is a flat line at 0mV on the time domain graph. The mid (lime green, 800Hz-3kHz) was boosted to only 0.8 times the max which can be seen in Figure 15 and 16 having a magnitude of approximately 0.8 times that of the dark green trace. In the time domain, the 1Vpp input is boosted to about 3.2Vpp which also aligns with the theory of it being 0.8 times the max volume. The dark green trace is the high-mid (3-10kHz), which was boosted to the max and can be seen in Figure 15 and 16. With an input voltage of about 1Vpp, it has been boosted to about 4Vpp, which is to be expected given the speaker's impedance. Lastly, the treble (pink, 10kHz+) was also set to the max and can be seen in both Figure 15 and 16 that it is steadily growing as the signal frequency increases. It is expected that if this frequency continued to grow, the treble output voltage and frequency magnitude would also continue to grow as well. Although the overall gain is hard to say for certain as a result of fluctuations in the input signal, the trend suggests that the voltage should not exceed 5V, which is the max, aligning with the design of the equaliser in section 4.2 Module Design and Discussion, in ensuring the speaker operates within its power ratings.

In terms of limitations of the simulations, on top of the cumulative limitations mentioned for each individual module, there is a limitation in the audio file chosen. All the previous values, although they are averages at a number of different points using cursors, they are only approximations very close to the true value as a result of fluctuations within the input signal that occurred due to compressing the original 35 second audio file into only 3 seconds, which introduced significant amounts of noise and noticeable distortions in the input signal, especially towards the beginning and end. Moreover, for all the simulations as a whole, the results are limited due to the component specifications. Many of the components like the LM386 power amplifier, and the LM4562 were sourced from the internet. After reading the files associated with these components, they were compared to the appropriate datasheets to ensure accurate simulation. However, the majority of the values are rounded to 3 decimal places, which may introduce cumulative small decimal errors during simulation time. Another discrepancy is that the frequencies over 10kHz were also cut during the compression of the audio file (this can be seen in Figure 8 of the input stage where there is no signal). As a result of this, the magnitude for the frequencies over 10kHz seem to either be missing or smaller than expected within every simulation result. With that being said, the simulations demonstrate the chosen design's capabilities and functionality as an audio crossover network with an integrated 5-band equaliser while maintaining the shape of the original signal, with respect to impedance and power handling.

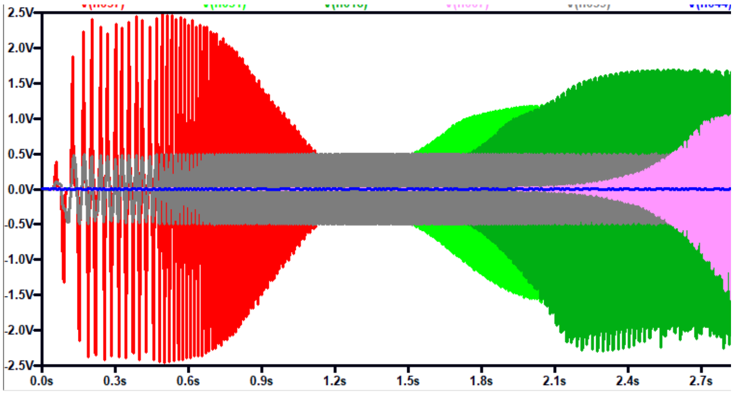


Figure 15: Time Domain Simulation of entire Audio Network

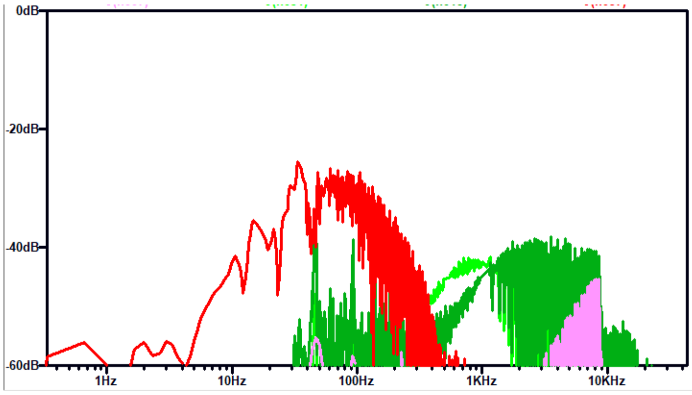


Figure 16: Frequency Response of Entire Audio Network

6. Prototyping and Iterative Designs

6.1 Implementation Process and Prototyping Discussion

After meeting with the client and reviewing feedback, it has been decided to move forward with the chosen design. Using a systematic approach, each module is first independently physically implemented, like in figure 17, using its appropriate design and components based on the above schematics to ensure that the modules' design performs as expected and ease maintenance as well as troubleshooting issues that may arise before combining all the modules into a finalised prototype. They are then tested which will be explored in section *7. Testing, Analysis and Verification*. Interestingly, the initial design from the preliminary report actually worked decently well, so the final design does not dramatically change too much from the initial proposed design. Throughout the prototyping phase, a number of factors were considered. In terms of capabilities, on top of performance, cost, power consumption and other ideas previously explored in section *4. Design Overview and Evidence*, overall circuit layout, and to a further extent, aesthetics were also heavily considered. Moreover, user friendliness was at the forefront so iterative designs heavily considered how usable the system is. This is vital since it comes together into a presentable and visually appealing product. However, technical challenges and limitations were encountered during this phase, which will be explored in the section below. Through many iterative design changes and tests, these issues were addressed. Additionally, future renditions will make use of an allpass filter to decrease the destructive interference that occurs, resulting in signal loss due to the phase differences between the filters. This cannot be implemented for the time being due to a limited number of audio op amps in the electrical kit, but most notably the additional power demands that are required on top of the already high power design (AD2 overcurrent condition appears due to the increased power demand). The issue is very minimal and the slight boost in performance does not justify the additional costly requirements.

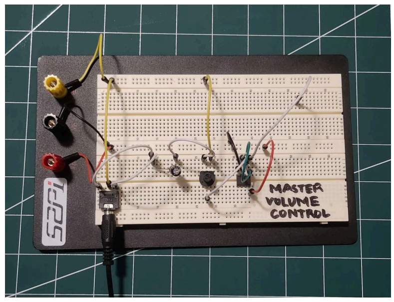


Figure 17: Early Prototype of Input Stage

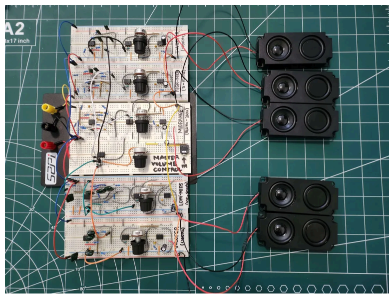


Figure 18: Initial Prototype of Audio Crossover and Equaliser Network (Enlarged ver. in Appendix A)

6.2 Iterative Designs and Ideas

Below is a table with all the iterative design changes that occurred in the prototyping/testing phase with the issues and reasons for these changes. Test results for the finalised design with the iterative changes and discussions are in the following section.

Iterative Change Implemented	**Issue and Rationale for Change**
Changed most Dupont wires to U shaped jumper wires	The large amount of wires overlapping and making contact with each other and other components caused signal interference which led to erratic behavior, and unintended connections between parts of the circuit. It also made it hard to troubleshoot and maintain. The change to U shaped wires not only solved these issues but also made the layout more aesthetic, allowing it to be easier to turn the potentiometers without contacting the wire, potentially affecting their connection to the rest of the circuit.
Added knobs to better quality potentiometers	The higher quality potentiometer allows for better grounding and ease of use due to having not only a more functional and visually appealing volume control which allows for better grip and turning capabilities, but also overall performance. A tradeoff is that they take up more room on the breadboard with the added knobs as well as facing sideways instead of straight up.
Circuit implemented across 6 breadboards	The modules being too close together was not very user friendly since all the volume controls were clumped together. It also further increased the risk of short circuits from pins unintentionally touching each other, possible thermal buildup which could affect the many op amps performance, difficulty in measuring results, troubleshooting and maintenance, and even possible electromagnetic interference especially at high frequencies. Hence, each filter/band was given a dedicated breadboard as well as the input stage. This has the tradeoff of increasing the total resistance of the network, potentially affecting the output signal.

Removed 5 Buffer Amps	The circuit required way too much power so the 5 buffer amps between the crossover and equaliser were removed to reduce power consumption. After testing, the removal of these buffer amps had minimal effect on the performance of the network.
Removed Zobel Network from equaliser	The Zobel network was eliminated despite it theoretically helping with impedance matching of the speakers as well as improve the performance of the 5 way audio crossover. In practice, it made little impact to the overall sound quality and crossover behaviour, while notably suppressed the high frequency signals. It only added a layer of additional cost to the circuit.
Increase gain of tweeter	Since the high frequencies were cut in the simulations, it went unnoticed how small the magnitude for the tweeter actually was. Hence, the decision was made to add a $10\mu F$ capacitor to the LM386 power amp between pin 1 and 8, to achieve 200 gain.
Use 5 speakers instead of 3	It was decided that 5 speakers would be used instead of 3 so summing amplifiers would not be needed, increasing the power demands of the circuit (overcurrent condition). The 5 speakers also allow for testing edge cases with low and high frequencies.
Increase Vcc and Vee	It was noticed that the previously chosen $\pm\ 5V$ for Vcc and Vee did not allow for enough headroom so the signal would saturate quite frequently, leading to a 'popping' noise. So these values have been increased to 6V (Vcc) and -6V (Vee) in order to prevent saturation while still being within the speaker's power ratings. This removed the aforementioned 'popping' noise.

6.3 Finalised Prototype Design

After a number of iterations of the chosen design, throughout the prototyping and testing phase, of the audio crossover and 5 band equaliser network, below in Figure 19 is a finalised version of the schematic of the entire design and Figure 20 is a finalised version of the entire prototype design. This prototype is vigorously tested, analysed, and verified in section *7. Testing, Analysis and Verification*, along with discussions, discrepancies and justifications. Overall, to achieve this finalised prototype, including input stage, crossover, equaliser, and output stage as well as the desired stretch capabilities, adjustments based on frequency response, audio performance, power requirements, and user feedback, was utilised and trialed through numerous tests. A decision was also made to mount the entire circuit on a black wooden board and zip tie the speakers onto an acrylic sheet for aesthetics as well as robustness of the audio network for the live demo. The final product, a presentable audio crossover and equaliser network, is a result of numerous iterations and design changes as can be seen in the table above.

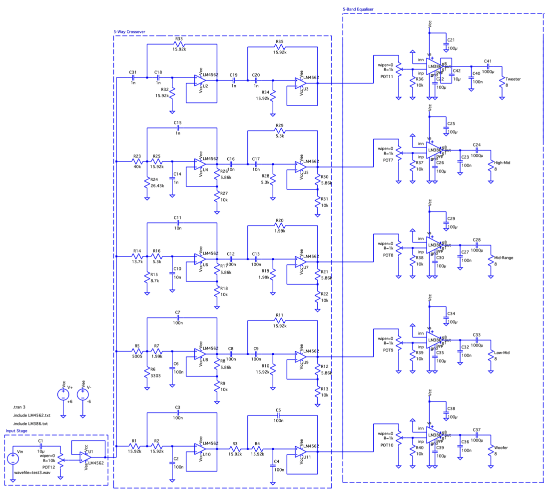


Figure 19: Schematic of Audio Crossover and Equaliser Network with the iterative changes implemented (Enlarged ver. in Appendix A)

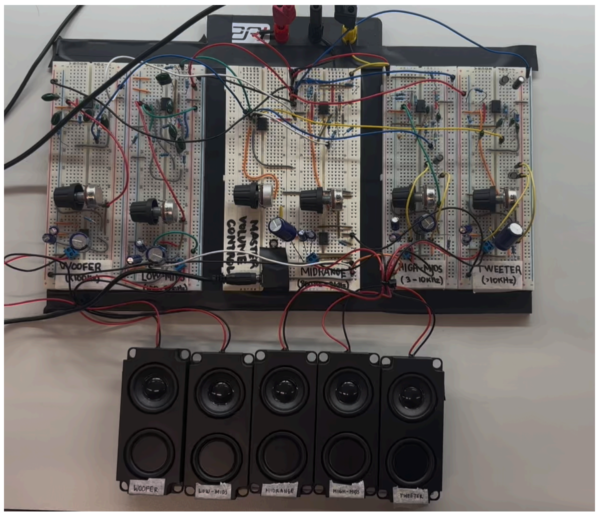


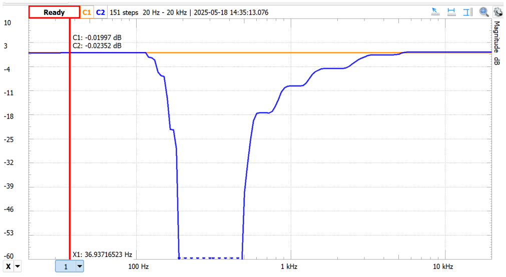
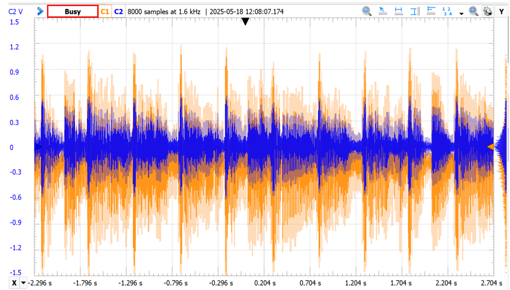
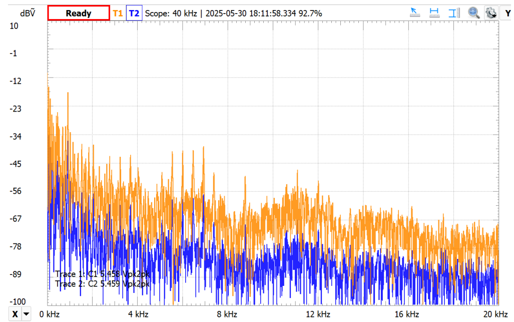
Figure 20: Finalised prototype of Audio Crossover and Equaliser Network used in the presentation and live demo (Enlarged ver. in Appendix A)

7. Testing, Analysis and Verification

The design of the testing procedure saw each module having a minimum of two tests to ensure that it is reliable and repeatable, working under different operating conditions. The first test is a frequency sweep using the AD2 network analyser ranging from 20Hz to 20kHz to match the simulation and cover the human audible spectrum, in order to observe the module's frequency response. The second and third test is a song, (Paramore's *Hard Times*, a song with large frequency ranges) to demonstrate the modules' capabilities, effectiveness, and performance under practical application (this was recorded using scope for time domain and the spectrum analyser for frequency domain. However, the spectrum analyser files got corrupted before the presentation. The circuit was water damaged by the rain whilst getting transported home, and there is not enough time or components to rebuild the entire circuit. Only the input stage and the equaliser and output stage were able to be salvaged so the spectrum analyser was only retested for these two modules, unfortunately). These will be conducted under different conditions (volume levels) to simulate real world application. The overall design of the audio crossover and 5 band equaliser is then physically implemented and tested the same way to ensure reliability and repeatability of the design, following similar test conditions as the modules. Measurements will be taken with the AD2 to provide the most standardised results possible at different frequency ranges for the audio crossover and equaliser performance. Through iteration, a finalised project with adjustments based on frequency response, audio performance, power requirements, and user feedback will be presented. They will then all be analysed and verified through their discussions, with any discrepancies justified. Enlarged versions of all the different testing results will be placed within *Appendix A: Enlarged Higher Resolution Figures*.

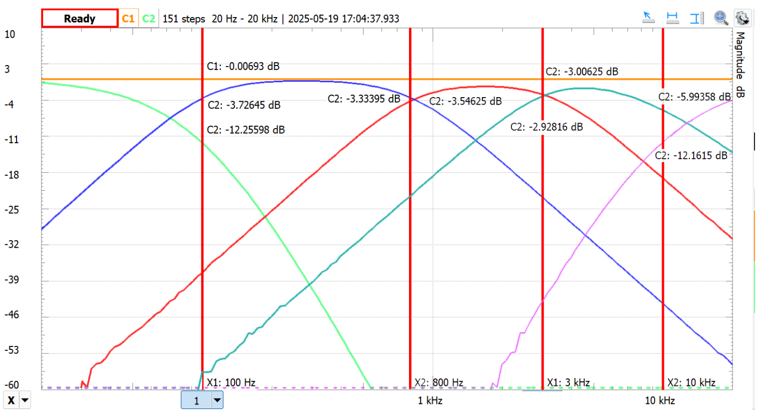
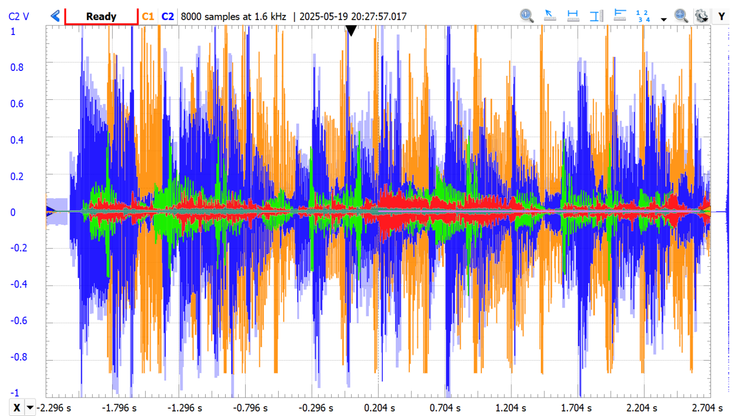
7.1 Testing and Analysis: Modules

Module 1: Audio Input Stage

Test Number and Testing Parameters		
Test 1: Frequency Domain Input: Frequency Sweep Volume Setting: 1. Start at MAX 2. Decrease to MIN 3. Increase to MAX	**Test 2:** Time Domain Input: Song Audio Volume Setting: $0.4 \times$ MAX	**Test 3:** Frequency Domain Input: Song Audio Volume Setting: $0.4 \times$ MAX
Oscilloscope Results (Orange Input, Blue Output)		
 *Figure 21: Experimental Frequency Response of Input Stage*	 *Figure 22: Experimental Input Stage with Song Audio input (Time Domain)*	 *Figure 23: Experimental Input Stage with Song Audio input (Frequency Domain)*

These results are exactly as expected, aligning with the theory. In test 1, the volume was initially at max and as the potentiometer is turned to increase the resistance, the magnitude of the output signal begins to decrease until it completely flatlines, when the potentiometer is grounded, effectively muting the input signal. This demonstrates its capabilities in being a master volume control. In test 2 and 3, the master volume is set to 0.4 times the max, and it can be seen that the output signal is roughly 0.4 times the magnitude of the input signal (not exactly due to physical limitations with the trimpot). It is able to change the magnitude of the input signal while maintaining its relative shape as seen in figure 22 and 23. The bass signals may seem a little smaller relative to other frequencies, due to the DC blocking capacitor having a high pass filter like behaviour. Moreover, when the potentiometer is turned to the max, it is only able to reach the same magnitude as the input signal, confirming the limitation that it is only able to decrease the volume, rather than increase it. Notably, looking at the cursor in test 1, when the potentiometer is max, it is unable to completely match the magnitude of the input signal; it is ever so slightly lower (percentage discrepancy of 0.178%). This minor discrepancy stems from the fact that compared to the LTSpice which simulates ideal conditions, in reality, there are many non-ideal characteristics that justify these discrepancies which will be covered in section [8. Discussion and Justification of Overall Results.]

Module 2: 5-Way Active Audio Crossover Filters

Test Number and Testing Parameters	
Test 1: Frequency Domain Input: Frequency Sweep Volume Setting: N/A	**Test 2:** Time Domain Input: Song Audio Volume Setting: N/A
Oscilloscope Results	
 *Figure 24: Frequency Response of 5 Way Crossover (Experimental Bode Plots)*	 *Figure 25: Experimental Crossover with Song Audio Input*

These results performed as expected, largely aligning with the theory discussed but with a few discrepancies. In test 1, a 1V power supply was used to obtain the frequency response of the 5-way crossover using the network analyser. As shown in Figure 24, the resulting bode plot showcases how the crossover successfully splits the input signal into five distinct frequency bands ranging from 20 Hz to 20k Hz. The crossover points were clearly visible and corresponded well with the filter characteristics. Similarly, in the time-domain analysis in Figure 25, the crossover stage responds dynamically, with different filter stages activating depending on the frequency content of the song.

When examining the experimental bode plot, the aim was to achieve a nominal gain of 1 for the bandpass filters by implementing a voltage divider configuration, placing the peak of each crossover at 0 dB. However, this was not entirely achieved across all outputs. In the mid-range and high-mids bands, the actual peak gain deviated slightly below the expected 0 dB, but this did not have a noticeable difference when tested with real world song audio inputs. These discrepancies stem from the limitations in component availability, such as needing to substitute ideal calculated resistor values with available E12 standard series resistors at Telstra Creator Space. The effect of resistor and capacitor tolerances, ±5% and ±10% respectively, also introduces gain inconsistencies and slight frequency shifts compared to their theoretical cutoffs.

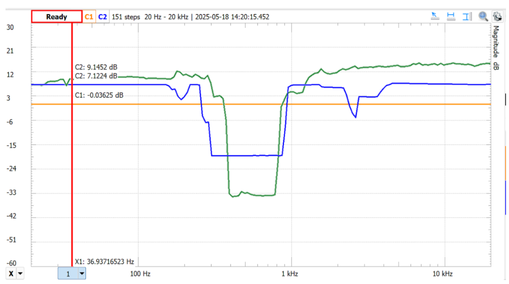
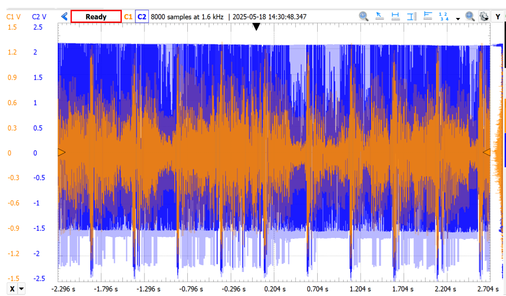
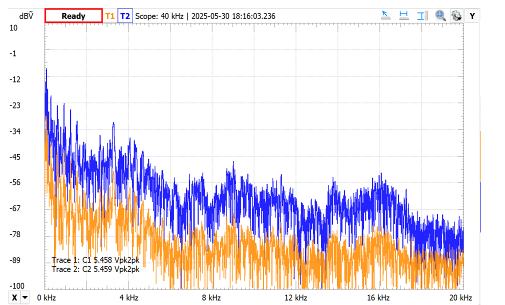
When examining the experimental crossover with song audio input, there are very big discrepancies between the magnitude of the different frequency bands, namely the high-mids and tweeter. This is partially due to the analogue input being a very short snippet of the song, which removed a lot of the higher frequencies during compression and introduced a lot of noise into the signal, which is why the magnitude of the high-mids and tweeter appears particularly small. Furthermore, destructive interference due to phase misalignment across filter outputs also contributed to this signal loss, causing significant volume loss in these bands. This volume loss will be mitigated in the equaliser section by adding a gain to the output signal for the tweeter. The chosen song, *Hard Times* by Paramore, consists of frequency content primarily in the low-mids range, which explains its dominating magnitude in Figure 25. It also happens to be around the average frequency for a female voice.

To quantify the discrepancies between theoretical and experimental cut off values and roll off rates, percentage errors were calculated and are outlined in the table to the right. The graph in which these values are obtained are attached in [Appendix A](#) due to over cluttering. As expected, most filters performed within an acceptable margin, especially the woofer and tweeter ranges, which used fourth order filters and thus provided more precise roll-off characteristics. The woofer low-pass filter showed a 2.17% error in gain at cutoff and only an 11.67% error in roll-off rate, while the tweeter high-pass was even closer to ideal, with just a 1.33% gain error and 10.92% roll-off rate error. These small deviations are consistent with the added sharpness of the fourth order Sallen-Key design, which is more robust against minor tolerance mismatches.

Frequency	Theoretical Frequency Cutoff	Experimental Frequency Cutoff	Frequency Cutoff Percentage Error	Theoretical Roll off Rate	Experimental Roll off Rate	Roll off Rate Percentage Error
100 Hz (Woofer)	- 12dB	-12.26dB	2.17%	-24dB/octave	-21.2 dB/octave	11.67%
100 Hz (Low-mids)	- 3dB	-3.72dB	24%	-12dB/octave	-9.86 dB/octave	17.83%
800 Hz	- 3 dB	-3.33dB	11%	-12dB/octave	-9.52 dB/octave	20.67%
3000 Hz	- 3dB	-2.93dB	2.33%	-12dB/octave	-8.12 dB/octave	32.33%
10,000 Hz (High-mid)	- 3dB	-5.99dB	99.67%	-12dB/octave	-8.71 dB/octave	27.42%
10,000 Hz (Tweeter)	- 12dB	- 12.16dB	1.33%	-24dB/octave	-21.38 dB/octave	10.92%

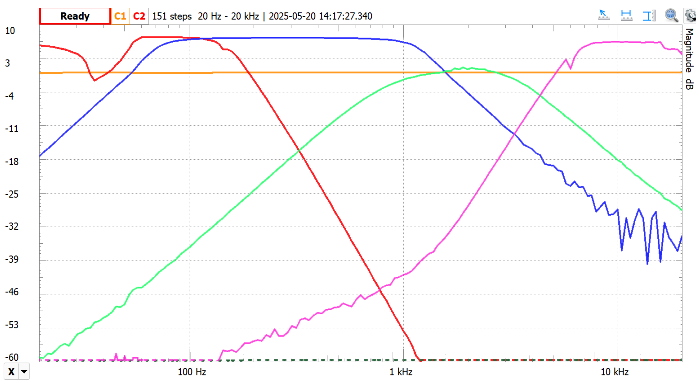
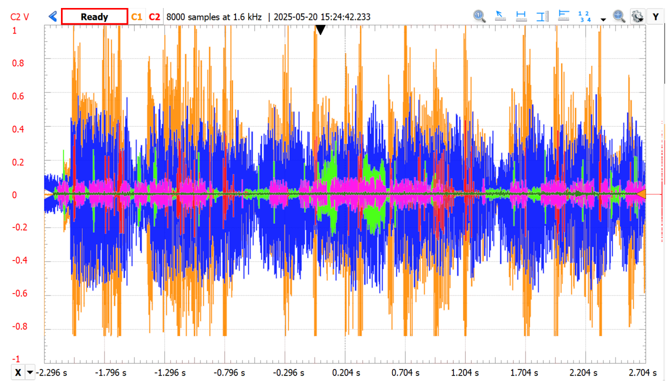
In contrast, the bandpass filters implemented second order filters and showed larger errors, particularly in the low-mid and high-mid bands. The high-mid filter at 10 kHz exhibited the most significant discrepancy, with a 99.67% cutoff error and a 27.42% error in roll-off, likely due to loose connections, or the circuit being built wrong, although it was thoroughly checked multiple times. More likely, the reason for this very large discrepancy, the LM4562 used for the high-mid was likely burnt out, since during testing, $\pm\ 20V$ was accidentally put through the op amp which is above its ratings[9]. This severely heated up the op amp, producing a white smoke. However, due to the limited number of components and the given timeframe, it was unable to be replaced right before the presentation. This will be an area of improvement for the future. Despite these issues being highlighted during network analyser testing, the crossover functioned as intended in practice when tested against several songs, though there was noticeable reduction in overall output volume which will be mitigated in the equaliser and output stage.

Module 3: 5-Band Audio Equaliser and Output Stage

Test Number and Testing Parameters		
Test 1: Frequency Domain Input: Frequency Sweep Volume Setting: 1. Start at MAX 2. Decrease to MIN 3. Increase to MAX	**Test 2:** Time Domain Input: Song Audio Volume Setting: $0.4 \times$ MAX	**Test 3:** Frequency Domain Input: Song Audio Volume Setting: $0.4 \times$ MAX
Oscilloscope Results (Orange Input, Blue and Green Output)		
 *Figure 26: Experimental Frequency Response of Equaliser and Output Stage*	 *Figure 27: Experimental Equaliser and Output Stage with Song Audio input (Time Domain)*	 *Figure 28: Experimental Equaliser and Output Stage with Song Audio input (Frequency Domain)*

The equaliser changed the most through iterative designs. In test 1, the green trace is the original equaliser while the blue trace is the iterated and refined equaliser. As can be seen, the old design is very unstable, producing a significant amount of noise and signal distortion compared to the new design which is considerably more stable. A notable tradeoff is that the new design is unable to get as loud as the old design with a lower gain as well as being able to mute and ground the signal more reliably. Regardless, as can be seen across all 3 tests, as the potentiometer is turned, the magnitude of the output becomes noticeably larger than the magnitude of the input while maintaining the original shape of the input signal. This validates this module as it is able to adjust the volume and amplitude of the input signal to a degree great enough in order to drive the speakers, and make the signal audible to the human ear, all while maintaining a power consumption that is below that rated of the speakers to prevent speaker damage. It is hard to say what the gain for the original equaliser design is due to how noisy the signal is, but for the redesigned equaliser, the gain matches that of the simulated value, with slight discrepancies due to the loading effects of the speakers and real world mechanical imperfections like input impedance, wire and breadboard resistance, and component capacitance and inductance. Interestingly, in test 1, there are noticeable dips as the potentiometer is turned. This discrepancy stems from the coupling capacitors at the output stage. Turning the volume knob causes a brief voltage surge as the capacitors have to charge and discharge, especially when also switching resistance paths. This may also stem from the wear in the potentiometers as well. In test 2 and 3, along with the input signal, unwanted noise is also amplified such as the input noise, and when the test song was compressed, it also introduced additional noise and signal distortion that this module also amplifies which is a limitation of this design. Despite having measures that block the unwanted low and high frequency noise, it is very difficult to perfectly guard against noise in general.

7.2 Testing and Analysis: Entire Crossover and Equaliser

Test Number and Testing Parameters	
Test 1: Frequency Domain Input: Frequency Sweep Volume Settings: - Mater Volume: MAX - Bass: MAX (Red) - Low-Mid: MAX (Dark Blue) - Mid: $0.85\times$ MAX (Light Green) - High-Mid: Muted (Dark Green) - Treble: MAX (Pink)	**Test 2:** Time Domain Input: Song Audio Volume Settings: - Mater Volume: 0.85 × MAX - Bass: MAX (Red) - Low-Mid: MAX (Dark Blue) - Mid: $0.85\times$ MAX (Light Green) - High-Mid: Muted (Dark Green) - Treble: MAX (Pink)
Oscilloscope Results	
 *Figure 29: Experimental Frequency Response of Combined Audio Circuit*	 *Figure 30: Experimental Combined Audio Circuit with Song Audio Input*

8. Discussion and Justification of Overall Results

The prototype has been iterated through many times, with adjustments implemented based on frequency response, audio performance, power requirements, and user feedback, which was utilised and trialed through numerous tests. The modular approach in conjunction with the iterative testing allowed for precise debugging of errors and easier optimisation of modules, before combining the different modules into a singular system, improving the reliability of the final integration. The entire system was then tested in section 7.2 Testing and Analysis: Entire Crossover and Equaliser. Based on these experimental results, it aligns well with the established theory. As can be seen with the different coloured traces, the crossover effectively separated the frequencies into their respective frequency bands, demonstrating the capabilities of the crossover. The magnitudes of these frequency bands were then able to be adjusted to the user's liking, as can be seen by the different magnitudes in figure 29, demonstrating the capabilities of the equaliser. Hence, all three modules work in conjunction. The testing parameters for the entire network were specifically chosen to demonstrate the functionality of the circuit.

The test parameters saw the master volume set to the max for the frequency domain and 0.85 times for the time domain, the bass (red) also set to the max, the low mid (dark blue) set to the max, the midrange (green) set to 0.85 times the max, the high mid (aqua) completely muted, and the treble also set to the max. These parameters hold up to the output results as in figure 29, the frequency response shows that the output signal is considerably larger than the input (orange) signal, which correlates to an increase in volume as expected. The bass, low-mid and treble also have the same magnitude as they are all set to the max volume. The midrange has a smaller magnitude as it was set to only 0.85 times the max volume, resulting in a magnitude that is roughly 0.85 times the other frequency bands. Notably, the high mid is completely missing from the frequency response graph. This is because it was completely muted so the signal was effectively grounded, as can be seen by the flat aqua line near the bottom of the graph in the frequency domain, and the flat line at 0mv in the time domain. The song frequently changes frequencies leading to the interesting result in the time domain in figure 30. It is also worth noting that the analogue input is only a snippet of a song that removed a lot of the higher frequencies during compression and introduced a lot of noise into the signal, which is why the magnitude of the tweeter and high mid appears particularly small. However, overall, this demonstrates the capabilities of the crossover and 5 band equaliser network as a whole, being able to adjust the magnitudes of the frequency bands to the user's liking.

However, there are a few interesting behaviours that arose across all the tests. Firstly, there is a bit of a dip in the bass of the frequency response before it levels out. This may be owing to the fact that this region is near the beginning of the frequency sweep signal. There are many capacitors throughout the crossover as well as the equaliser and output stage, whereby these capacitors take time in order to charge up, potentially resulting in this perceived dip. Moreover, the capacitor at the input stage has a high pass filter like behaviour, which may slightly attenuate these bass signals, resulting in the dip perceived. The low-midrange also seems exceptionally large in the time domain, Figure 30, and to a certain extent, also in the frequency domain. This is most likely due to the fact that the ω_{c2} for the low-midrange is considerably larger than ω_{c1}, when compared to the other frequency bands. Moreover, Paramore's *Hard Times* which is used in the testing of the analogue input primarily has frequencies that fall under the low-midrange frequencies compared to the others, resulting in a larger appearance on the experimental graphs. The treble (pink) would be significantly smaller in magnitude if a $10\mu F$ capacitor was not placed across pins 1 and 8 of the power amp of its respective equaliser. This resulted in a gain of 200, which greatly increased its magnitude in figure 30. It may also explain why the treble in the frequency domain, figure 20, appears to be less stable in comparison to the other bands.

A number of discrepancies were noted such as the loss of signal, slight noise and signal distortion. These stemmed from a few challenges and hurdles, which were eventually resolved, as described in section 6.2 *Iterative Designs and Ideas*. However, during the actual testing phase, these discrepancies were explored in greater depth. To begin, signal loss may have occurred due to the low quality audiojack frequently becoming loose, so it had to be taped down to the breadboard. Moreover, signal loss may have also occurred due to the additional resistances present in the circuit, mainly from the 6 breadboard used as well as the many jumper wires connecting everything together, as well as input impedance and component tolerances from the many resistors for example. There may also be unintended parasitic capacitance or inductance effects due to the wiring layout and breadboard connections. The many wires and components may have also had loose connections, resulting in distortion and signal loss. This was slightly alleviated with the switch to u shaped wires instead of dupont wires. The many interconnections have resistance, capacitance and other mechanical imperfections, with even possible poor grounding along with long wires introducing noise. The wear of the potentiometers may have also influenced potential loading issues. However, this issue was slightly rectified after hardware refinements like swapping to higher quality potentiometers. The circuit is high power consuming and it was noticed that when all the modules were close together (on 3 breadboards), heat was building up. The thermal effects on the many op amps could have potentially caused offset drift, bias current drift, thermal noise, changes to the slew rate and bandwidth, and other unideal gain and open loop characteristics. This issue was alleviated by spreading the system across multiple breadboards, improving thermal stability, reduced interference, and enhanced user access to controls, even though it slightly increased the overall resistance. There was also a distinct popping noise that would occur in practice. After increasing Vcc and Vee to $\pm\ 6V$, the issue disappeared entirely. This resulted from the headroom of the op amps saturating the signal, so by increasing the supply voltage, while maintaining the speaker's power ratings, this challenge was able to be overcome as well. There is also the discrepancy with signal loss that came from the destructive interference of overlapping bands due to the phase differences. This was covered in section 7.1 *Testing and Analysis: Modules*. As for the percentage errors that stemmed from experimental measurements, discrepancies may be a result of the accuracy tolerance of the oscilloscope used which has its own margin of error. Additionally, fluctuations or noise in the power supply can further affect the operation of the circuit.

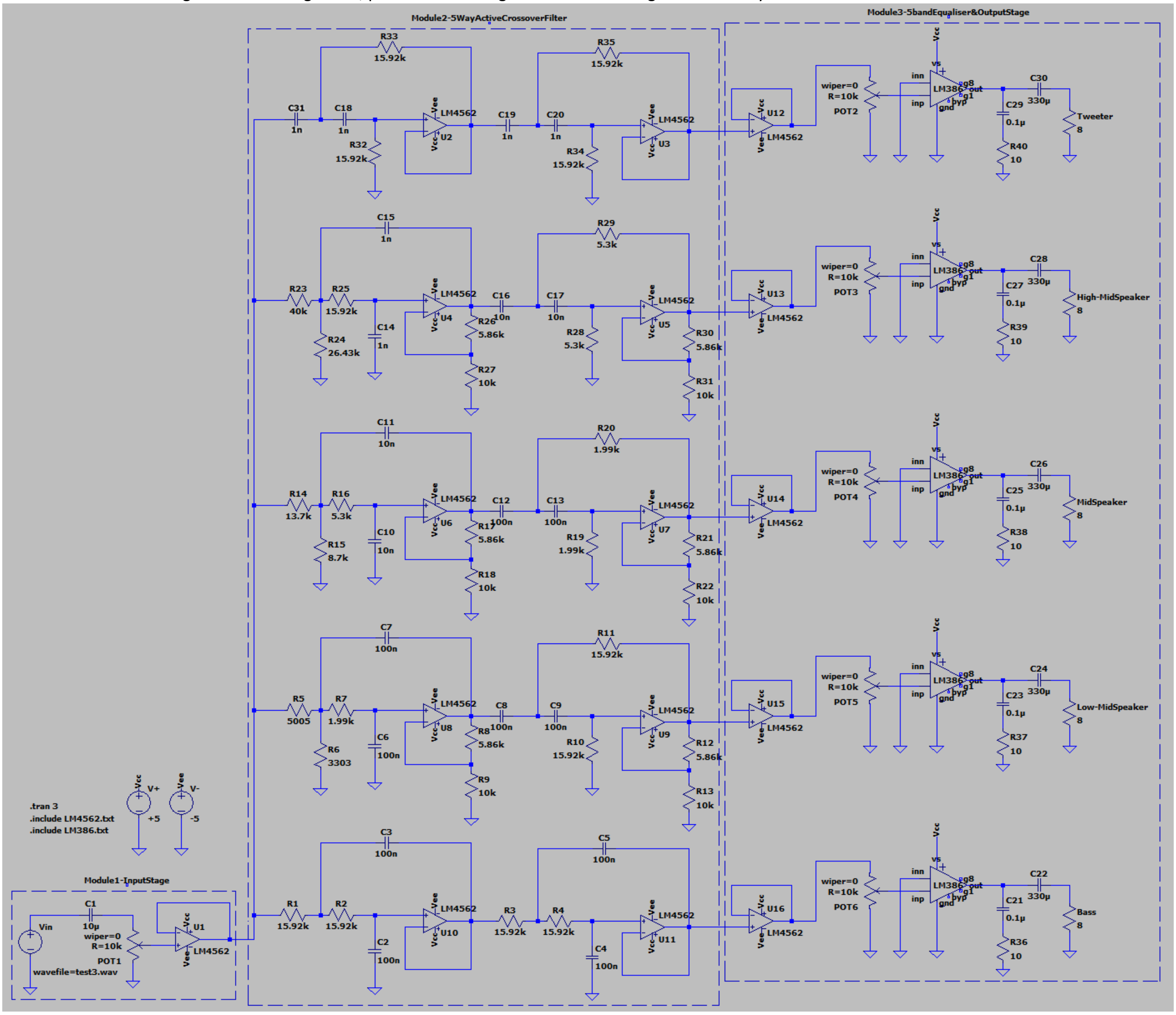
The circuit is high cost and high power consuming. However, this is justified by its reliable and stable performance. Additional cost and power requirements from the original design were cut by removing buffer amps and the Zobel network which reduced unnecessary cost and power demands without degrading system performance. This pragmatic approach preserved both AD2 and lab equipment functionality and focused on the most impactful design elements, whilst still working as an integrated equaliser audio system to a relatively high degree. While the design works well, with minimal distortion of the signal and high customisability, it still has room for future improvements. As the prototype has a master volume control and 5 band-specific volume controls, it is prudent to add some sort of visual LED display to keep track of each volume control potentiometer, as it is not currently user-friendly or intuitive to leave as simply knobs as it is difficult to visually see what setting it is on at any given time. Another improvement is the addition of all-pass filters following the crossover network in order to minimise the effects of destructive interference as mentioned above. This was considered to be added in the design but due to the power limitations and the time needed to fix such limitations, this was ultimately not added. A third improvement is the addition of a dynamic range control, which is another stretch capability. The final prototype does not have a dynamic range control, which means there is no automatic adjustment of signals that are too loud or soft. Thus, if signals are too loud, this may overdrive speakers and cause signal clipping. It may also damage components, heating them up and causing them to degrade faster. There was some research performed to implement one, using either compressor circuits to reduce the signal amplitude when it reaches a certain threshold or using limiter circuits to completely hard cap signals at a certain threshold. However, similar to the other improvements, due to time constraints, and the intention of getting one stretch goal working exceptionally well rather than multiple barely functional add-ons, time was fully invested on the 5 band equaliser, which was a necessary trade off.

9. Conclusion

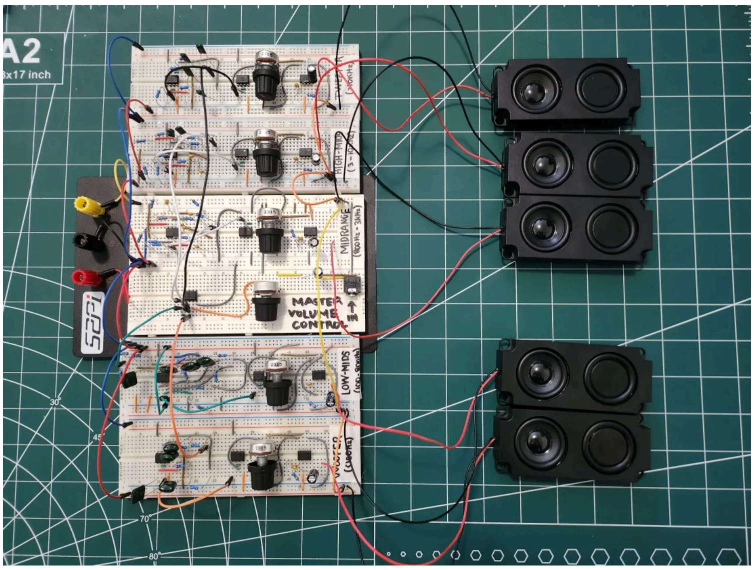
The project has successfully been implemented, with all the design capabilities being satisfied and demonstrated through simulation and rigorous testing. The final physical design - made up of the input stage, 5-way crossover network, equaliser and output stage - performs reliably, outputting clear, undistorted sound across the audible frequency spectrum. Each module was tested both individually and as part of an integrated system, meeting the performance requirements under different operating conditions (ie. different volumes). The input stage does not distort the signal and the master volume control attenuates the signal volume without issue. The signal then flows into the crossover which successfully splits the input signal into 5 frequency bands. Next, the 5-band equaliser modulates gain across the different frequency bands, allowing for precise adjustments and customisation of sound. The potentiometers also successfully allow for adjusting volume for each band and speaker. These signals then flow into their respective speakers without exceeding power limitations. Unwanted behaviours such as signal distortion due to signal clipping, phase misalignment causing destructive interference and unwanted noise were all mitigated through detailed and careful design iteration and component selection. Moreover, the system is organised and uncluttered, showing a dedication to aesthetics and user-friendliness, as well as actual performance. This outcome validates the success of the project, as all the design requirements are met.

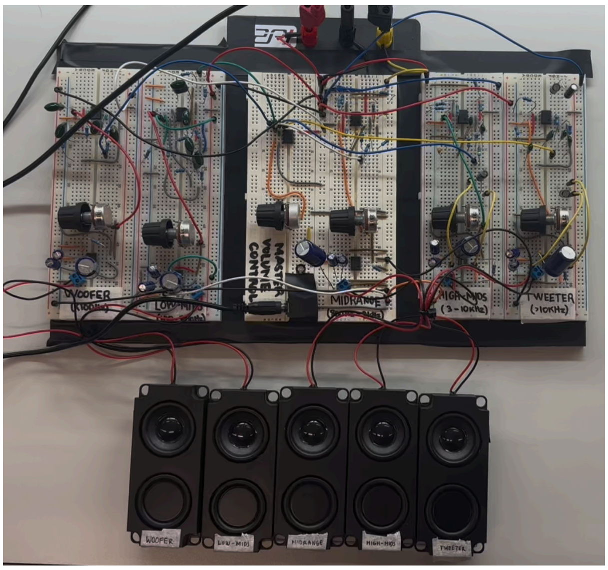
Appendix A: Enlarged Higher Resolution Figures

Below is an enlarged version of Figure 14, placed here for higher resolution and greater visibility.

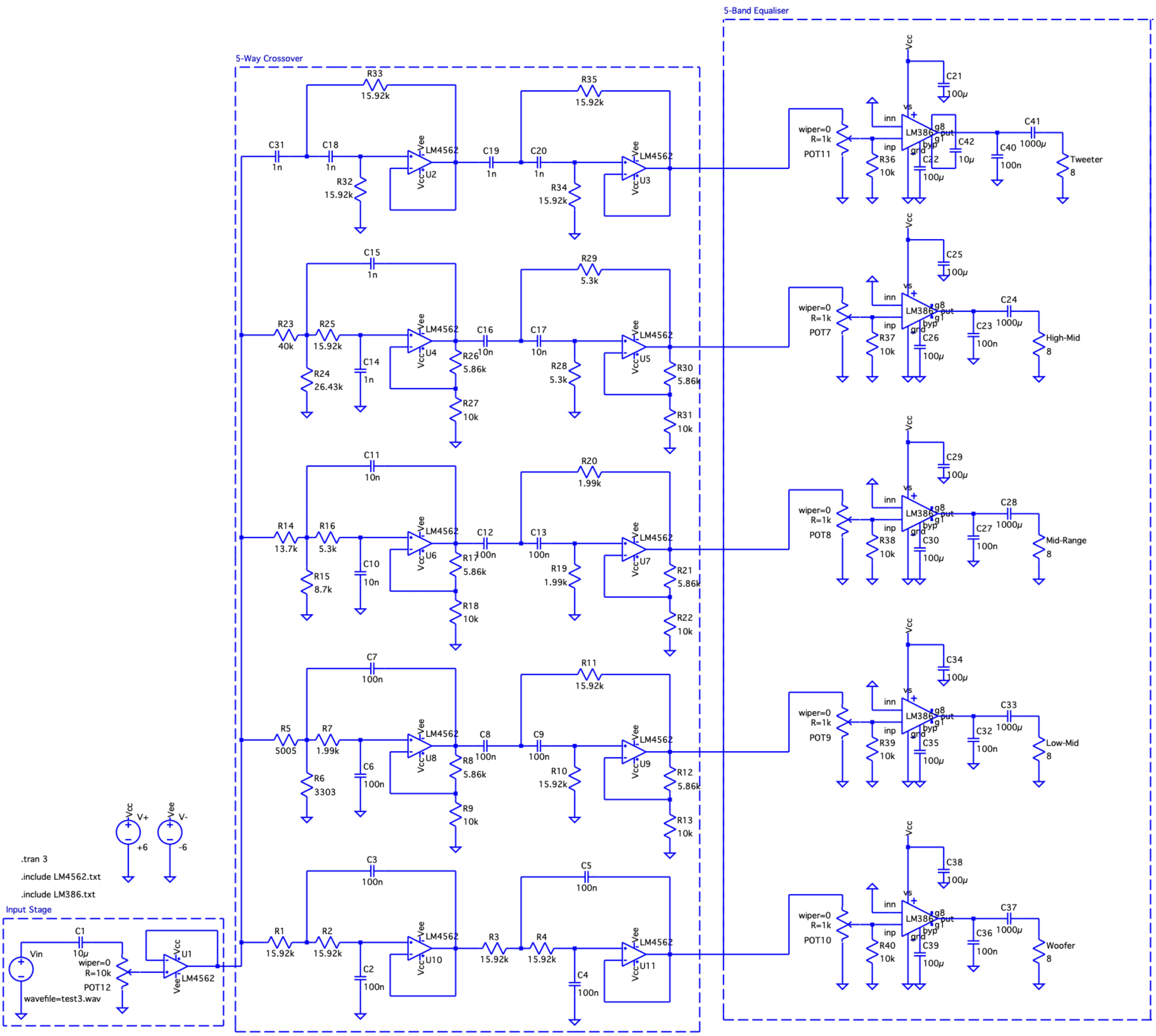


Below is an enlarged version of Figure 18 and 21, placed here for higher resolution and greater visibility.

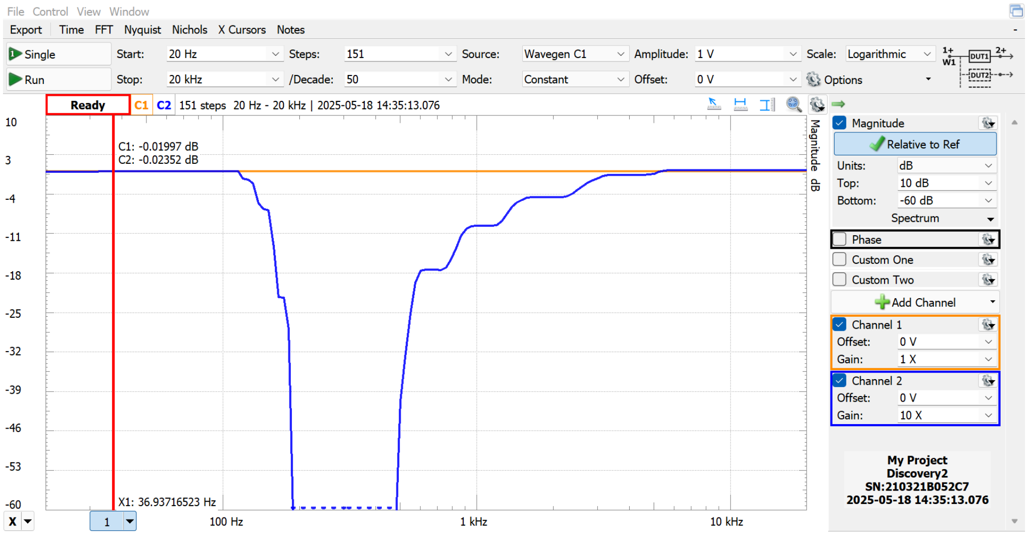


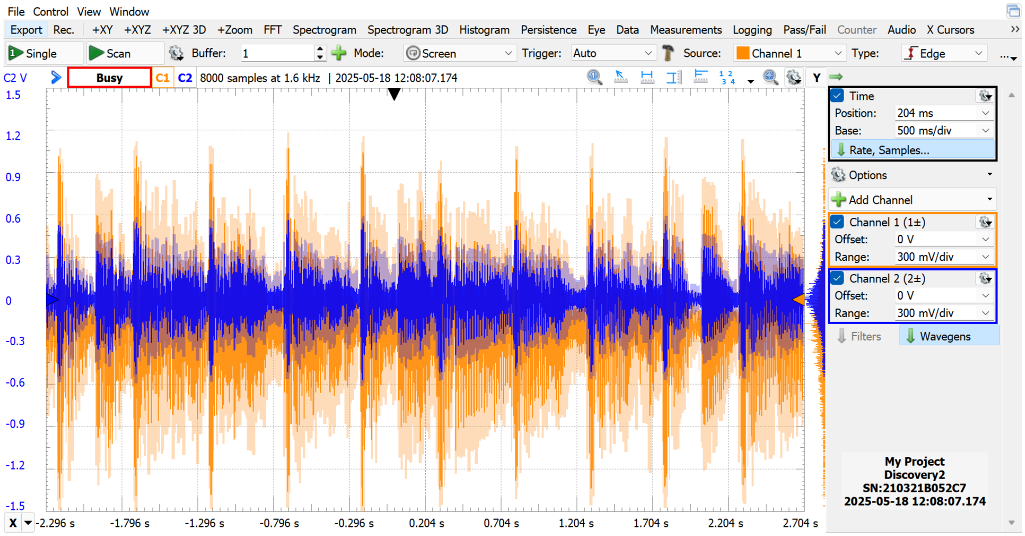


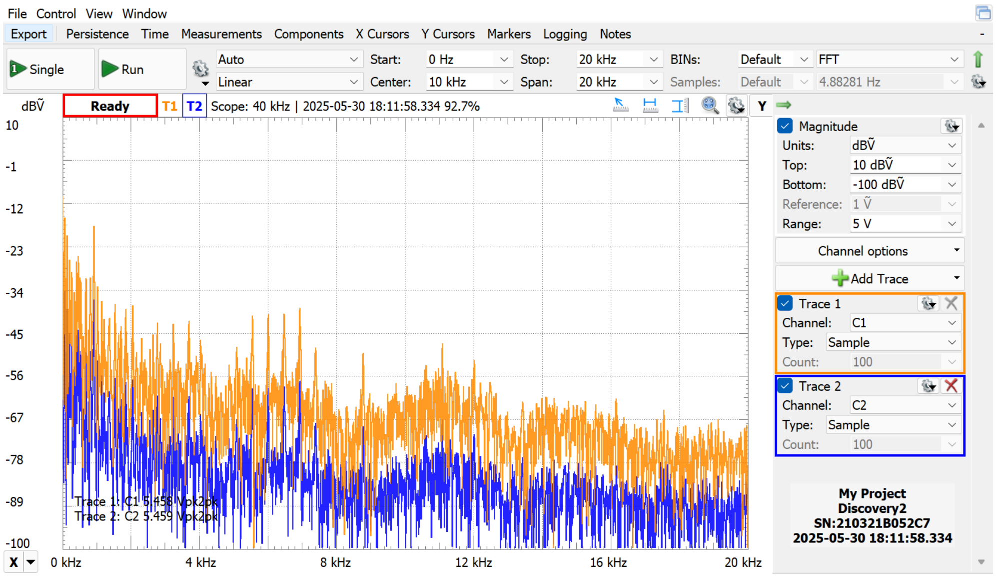
Below is an enlarged version of Figure 19, placed here for higher resolution and greater visibility.



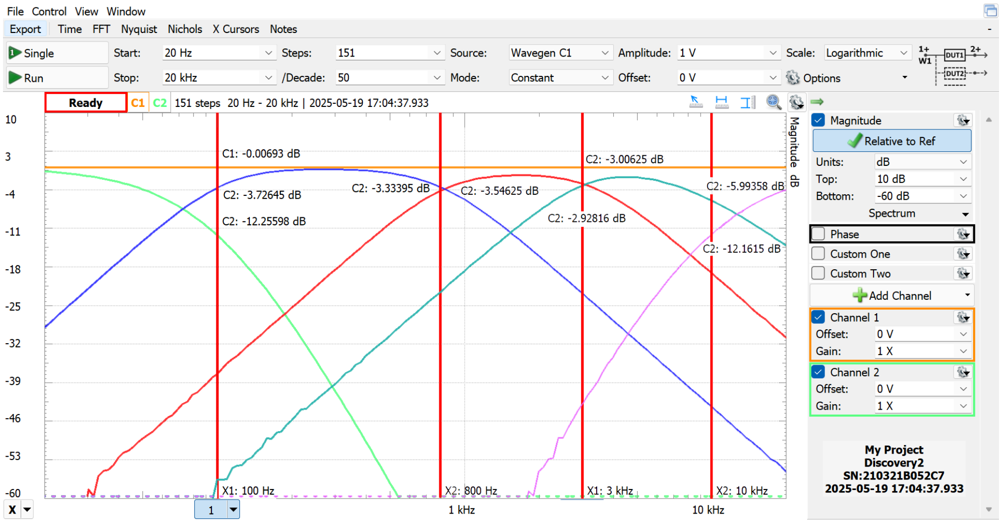
Below are the enlarged version of the test results for the input stage (Figure 21, 22, 23)

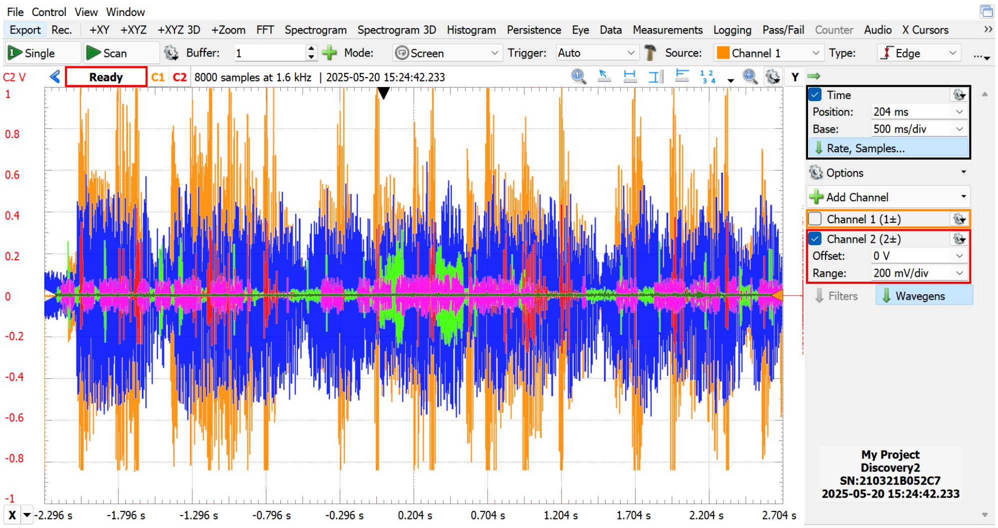




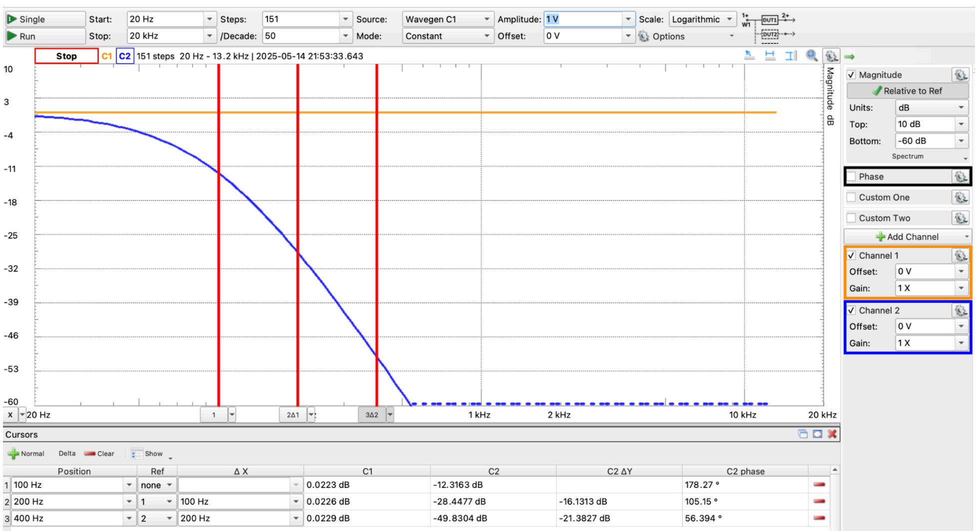


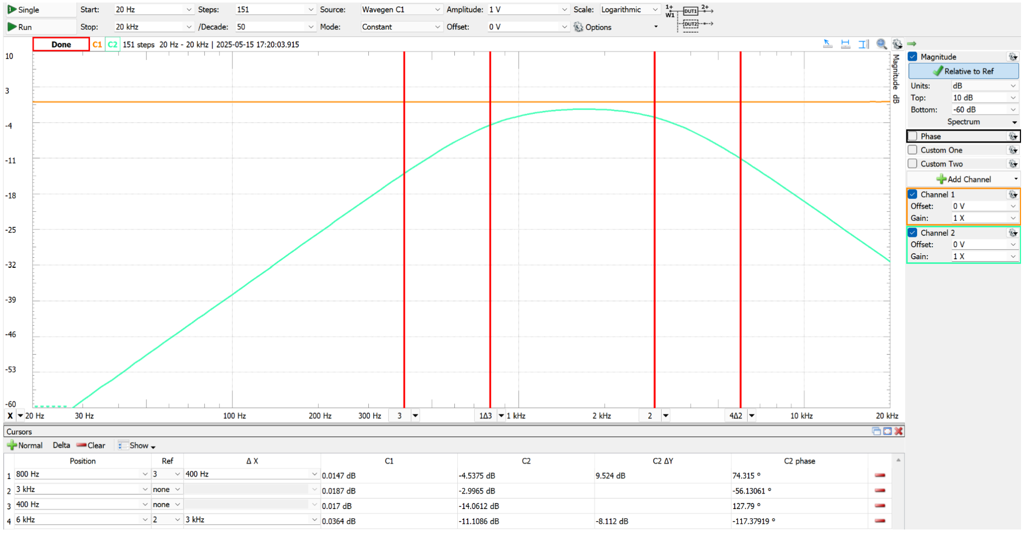
Below are the enlarged version of the test results for the crossover stage (Figure 24, 25)



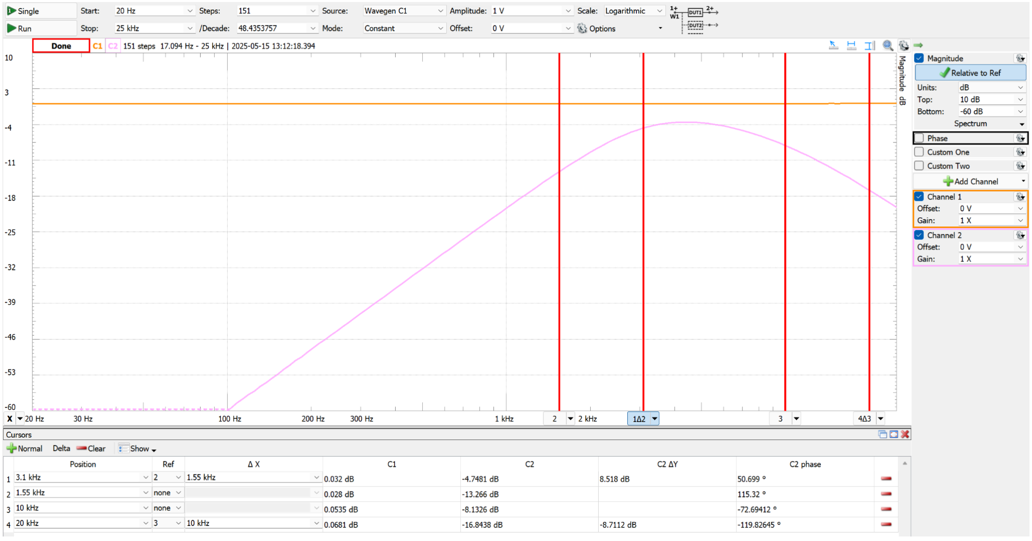


Below are the Figures used to calculate the roll-off rates in section 7.

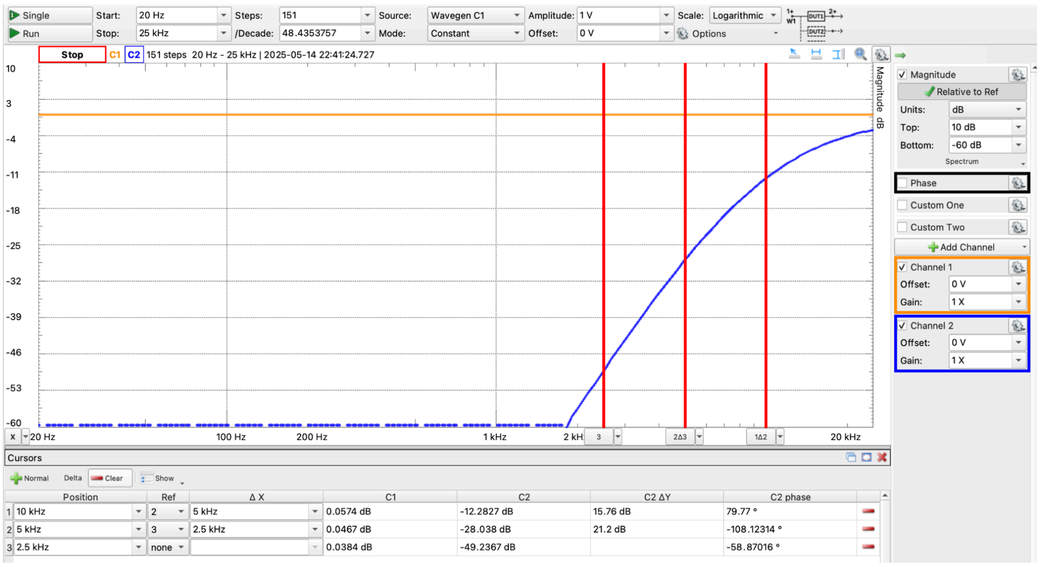




Single Start: 20 Hz Steps: 151 Source: Wavegen C1 Amplitude: 1 V Scale: Logarithmic
Run Stop: 20 kHz /Decade: 50 Mode: Constant Offset: 0 V Options
Done C1 C2 151 steps 20 Hz - 20 kHz | 2025-05-15 17:20:03.915
Magnitude
Relative to Ref
Units: dB
Top: 10 dB
Bottom: -60 dB
Spectrum
Phase
Custom One
Custom Two
Add Channel
Channel 1 Offset: 0 V Gain: 1 X
Channel 2 Offset: 0 V Gain: 1 X
Cursors
Position | Ref | Δ X | C1 | C2 | C2 ΔY | C2 phase
1 800 Hz | 3 | 400 Hz | 0.0147 dB | -4.5375 dB | 9.524 dB | 74.315 °
2 3 kHz | none | | 0.0187 dB | -2.9965 dB | | -56.13061 °
3 400 Hz | none | | 0.017 dB | -14.0612 dB | | 127.79 °
4 6 kHz | 2 | 3 kHz | 0.0364 dB | -11.1086 dB | -8.112 dB | -117.37919 °

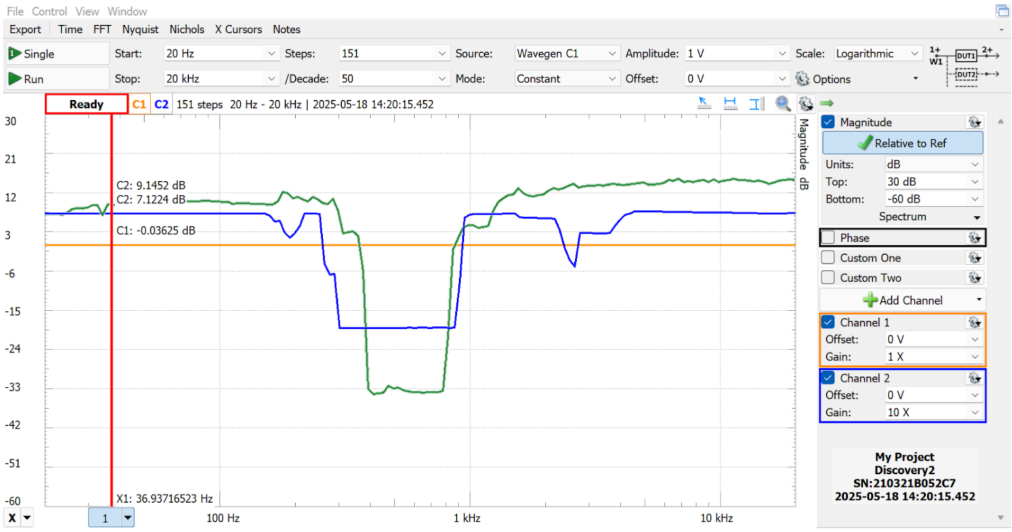


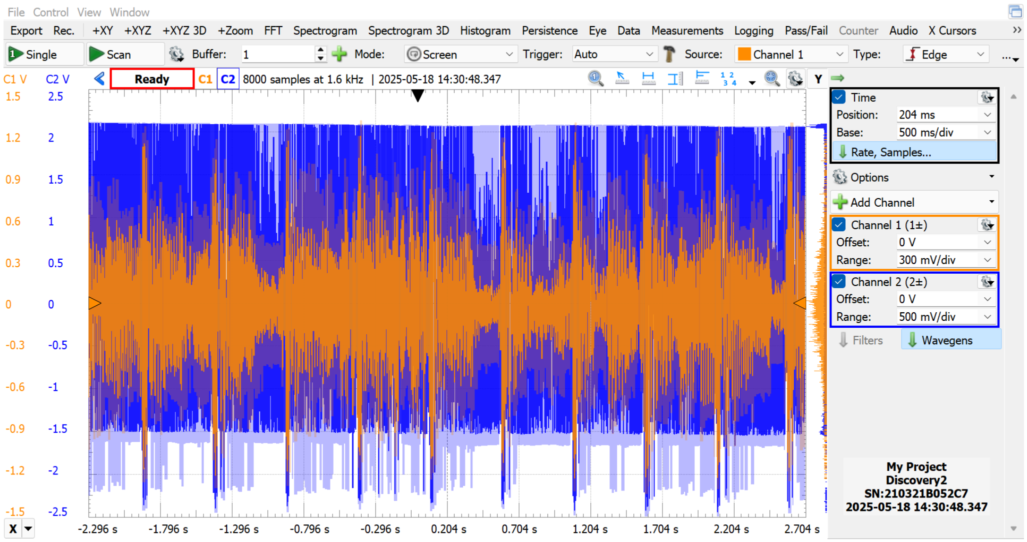
Single Start: 20 Hz Steps: 151 Source: Wavegen C1 Amplitude: 1 V Scale: Logarithmic
Run Stop: 25 kHz /Decade: 48.4353757 Mode: Constant Offset: 0 V Options
Done C1 C2 151 steps 17.094 Hz - 25 kHz | 2025-05-15 13:12:18.394
Cursors
Position | Ref | Δ X | C1 | C2 | C2 ΔY | C2 phase
1 3.1 kHz | 2 | 1.55 kHz | 0.032 dB | -4.7481 dB | 8.518 dB | 50.699 °
2 1.55 kHz | none | | 0.028 dB | -13.266 dB | | 115.32 °
3 10 kHz | none | | 0.0535 dB | -8.1326 dB | | -72.69412 °
4 20 kHz | 3 | 10 kHz | 0.0681 dB | -16.8438 dB | -8.7112 dB | -119.82645 °

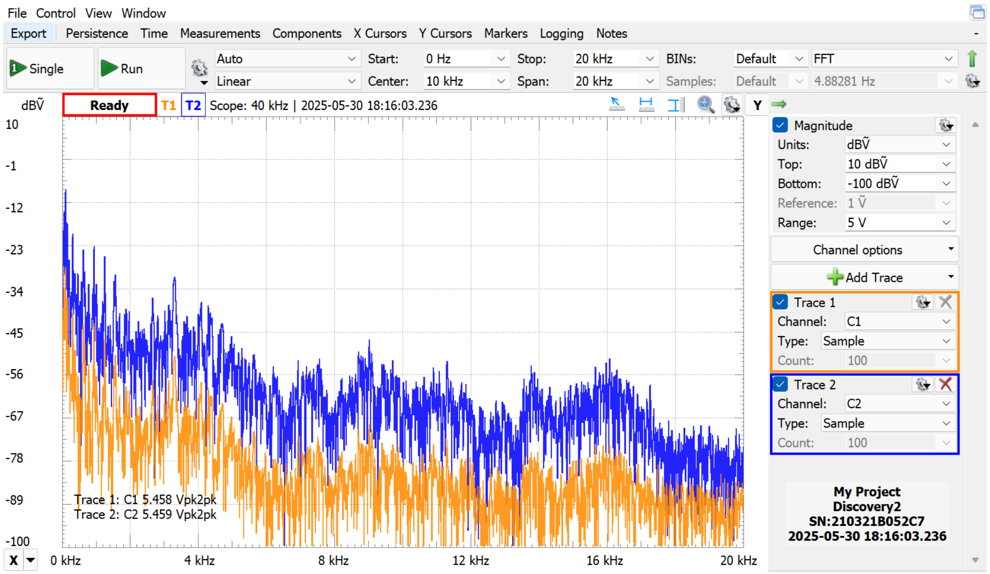


Single Start: 20 Hz Steps: 151 Source: Wavegen C1 Amplitude: 1 V Scale: Logarithmic
Run Stop: 25 kHz /Decade: 48.4353757 Mode: Constant Offset: 0 V Options
Stop C1 C2 151 steps 20 Hz - 25 kHz | 2025-05-14 22:41:24.727
Cursors
Position | Ref | Δ X | C1 | C2 | C2 ΔY | C2 phase
1 10 kHz | 2 | 5 kHz | 0.0574 dB | -12.2827 dB | 15.76 dB | 79.77 °
2 5 kHz | 3 | 2.5 kHz | 0.0467 dB | -28.038 dB | 21.2 dB | -108.12314 °
3 2.5 kHz | none | | 0.0384 dB | -49.2367 dB | | -58.87016 °

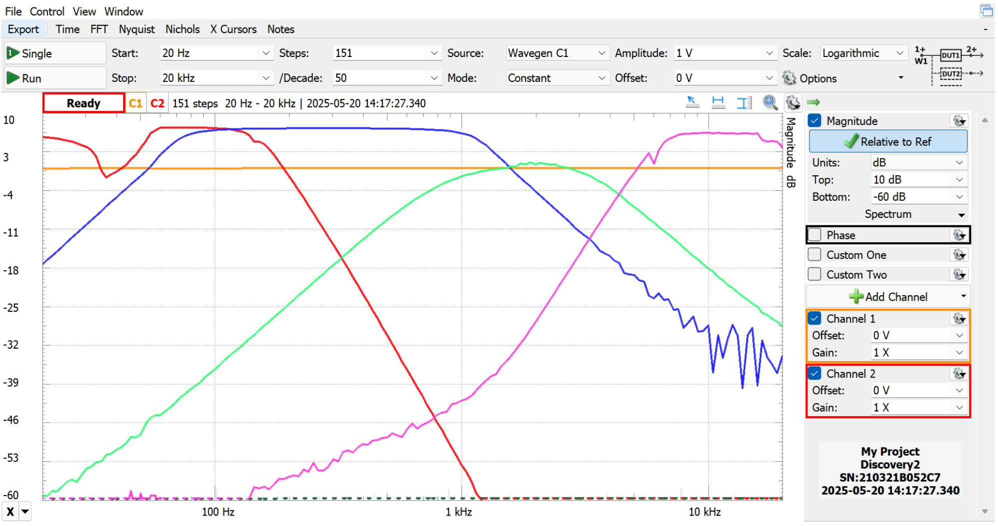
Below are the enlarged version of the test results for the equaliser and output stage (Figure 26, 27, 28)

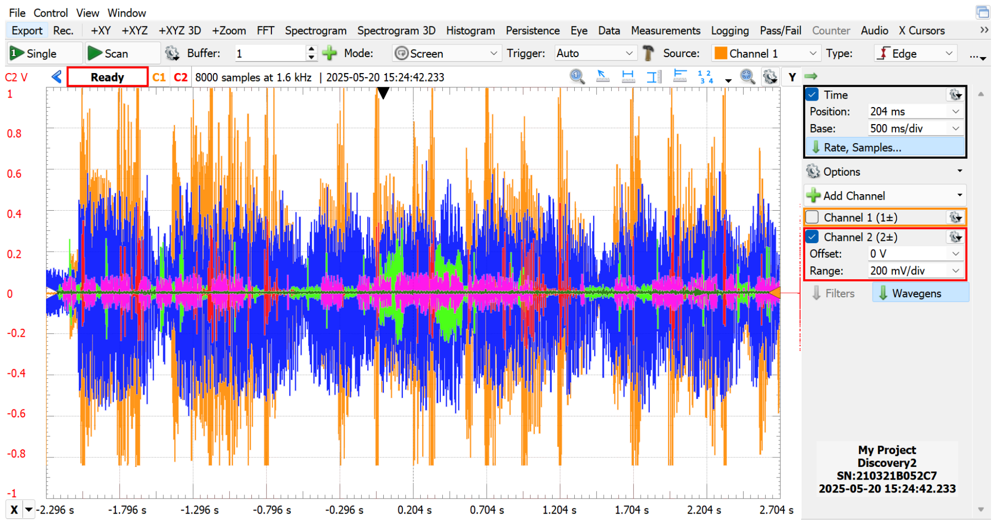






Below are the enlarged version of the test results for the entire audio crossover and equaliser network (Figure 29, 30).





Appendix B: References

The following references have been organised in the order of their in-text citation appearances.

Reference	In-Text Citation no.
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Appendix C: Contribution Statements

Below describes the work and tasks that each team member contributed in the making of this project report.

Team
As a team, we reviewed past literature in relation to audio crossover networks, and their associated power management circuits. We researched key principles of audio crossover design and audio equalisation with respect to acoustic properties and the typical frequency ranges of speakers, which allowed for the frequency ranges to be determined. The desired stretch feature was decided as a team. As a whole, the review, research, conceptualisation and project planning was done as a team effort. Everyone in the team communicated effectively, with planned project milestones, by attending regular self organised workshop sessions and meetings, and relayed necessary and important information via online communication methods. As a team, we finalised this preliminary report with the necessary edits and made it deliverable. As a team, we collaboratively prototyped the three respective modules, then tested the modules using the 3 testing parameters to ensure that it works and aligns with the theory, iterating the design when discrepancies arose to get it to match the theory and desired features even closer. We then collaboratively combined each individual prototyped module together in order to form an initial prototype of the entire audio crossover and equaliser network. This was done at the Telstra Creator Space. We noted challenges and issues that occurred with the prototype during iterative testing which led to many iterative designs of the entire circuit. We then worked together in order to form a presentation that displayed our project. We all contributed to the presentation media and routined a 2 minute live demonstration of the entire circuit's functionality. We attended weekly meetings and meetings before the oral presentation in order to rehearse our respective sections of the presentation. We also rehearsed our live demonstration as well at these meetings. All this cumulatively came together in the writing of this final report with the necessary edits and made it deliverable.

Tuong Bao Nguyen
I contributed to making the initial template for the project report. I contributed to the research of the audio input stage module which led to its final design, along with its associated discussion. I also chose the simulation test audio. I then simulated the input stage module using LTSpice, and discussed the results in comparison to the theory. I have also made a working prototype for this module using the AD2 and tested it. I wrote the methodology of the entire project report as well as the preliminary abstract, preliminary conclusion, overall simulation, overall simulation discussion and general design discussion. I then made the final edits necessary and submitted the preliminary project report for the team. I wrote the final abstract and contributed to the prototyping of the overall design, and selected all the testing parameters that displayed the functionality of the overall design. I conducted the test for all the modules as well as the combined circuit in its entirety in section 7.2. Testing and Analysis: Entire Equaliser and Crossover, ensuring that it works both with the AD2 as well as the lab equipment. I wrote the entirety of section 6. Prototyping and Iterative Designs, and their associated subsections. Then, I wrote the main discussion for section 7. Testing, Analysis and Verification. I also wrote the discussion for the testing of the input stage in section 7.1, analysing results, and providing any justifications for discrepancies. I partly wrote section 8, which is the main discussion and justifications for testing the entire circuit, and the project as a whole. I then made the final edits necessary and submitted the final project report for the team.

Signature:

Zhong Chun Xie

I contributed by first researching design principles of equalisers and the LM386 power amplifier. I designed the 5-band audio equaliser and output stage module, and its associated discussion, including impedance matching and power handling, while also simulating the module on LTSpice. I sourced files for the LTSpice simulations from the internet, notably for the LM4562 and LM386, and potentiometer. I also researched and wrote the introduction and conceptualisation sub section. I also helped in organising and leading meetings/workshops (organizing time and booking rooms, etc.).

I contributed to the prototyping of the overall design. I troubleshooted and iterated through the design of the equaliser, resulting in the final equaliser that was used in the presentation as well as the live demonstration. I also wrote the discussion for the testing of the equaliser and output stage in section 7.1, analysing results, and providing any justifications for discrepancies observed. I contributed to the writing of section 8, which relates to the main discussion and justifications of the overall project, and its testing. I also wrote the final report conclusion, section 9.

Signature:

Ayra Delfina

I contributed by first researching design principles of crossover networks as well as active filters. Using this knowledge, I designed the 5-way active audio crossover module, and its associated discussion. I then simulated the 5-way active audio crossover module and discussed the results in comparison to the theory. I also tested different audio inputs and analysed the crossover's performance. Moreover, I created the block diagram for the design finalisation. I was responsible for progress tracking, taking the meeting minutes at our regular meetings, and frequently updating our gantt chart. I also organised the references and in text citations.

I contributed to the prototyping of the overall design, notably all the sections of the crossover. I also wrote the discussion for the testing of the crossover in section 7.1, analysing results, and providing any justifications for discrepancies that were observed. I primarily aesthetically designed the slideshow for the presentation, made appropriate preparations, and formulated a demo routine for the preparation (selected a song and order of routine to showcase each equaliser band). I continued with the regular weekly meeting minutes, and continued to update the gantt chart as necessary. I also organised multiple meetings within the Telstra Creator Space with the intention of working on prototyping and testing the design.

Signature:

Appendix D: Demonstration of Audio Crossover and Equaliser

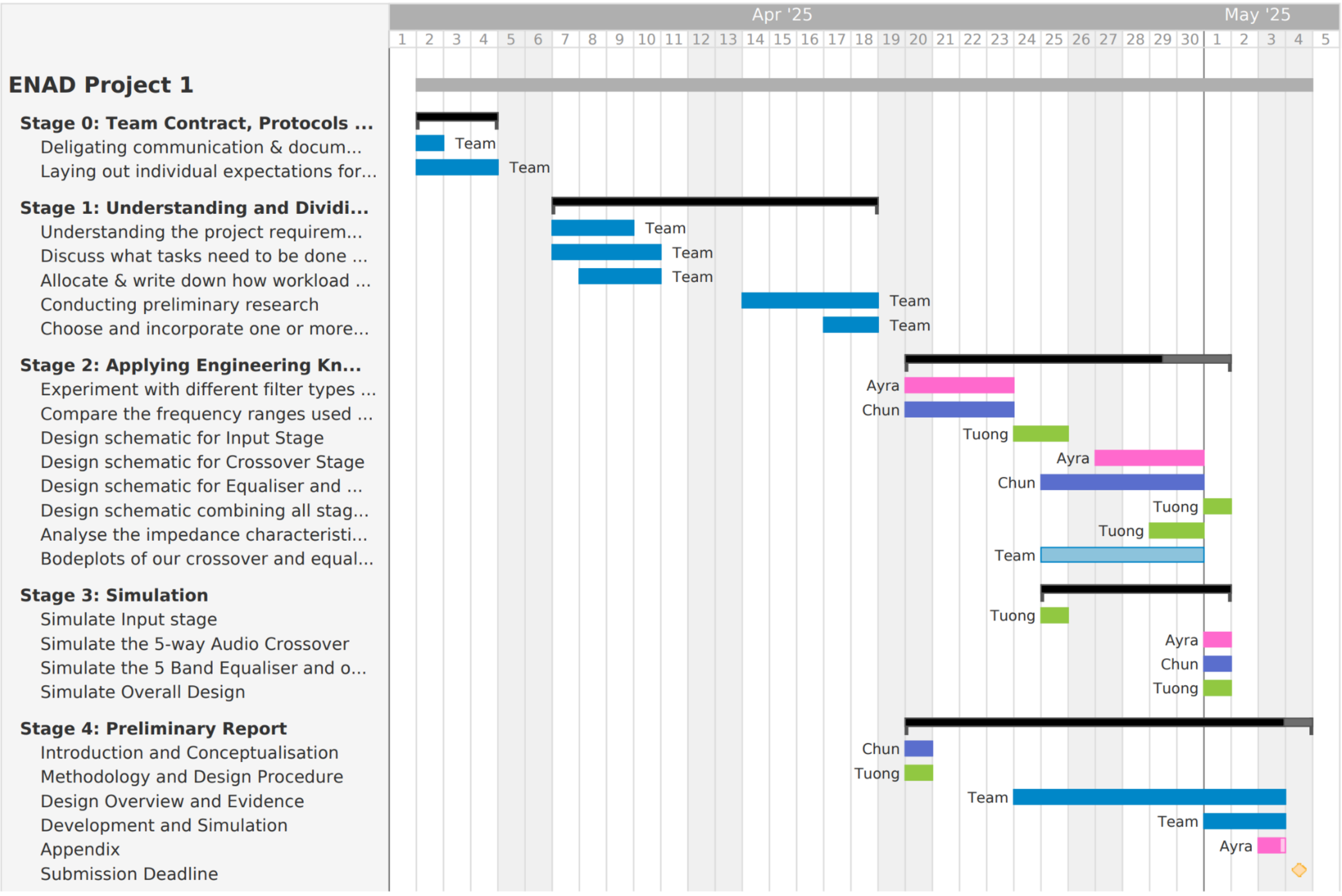
A video demonstrating the functionality of the circuit is attached below:

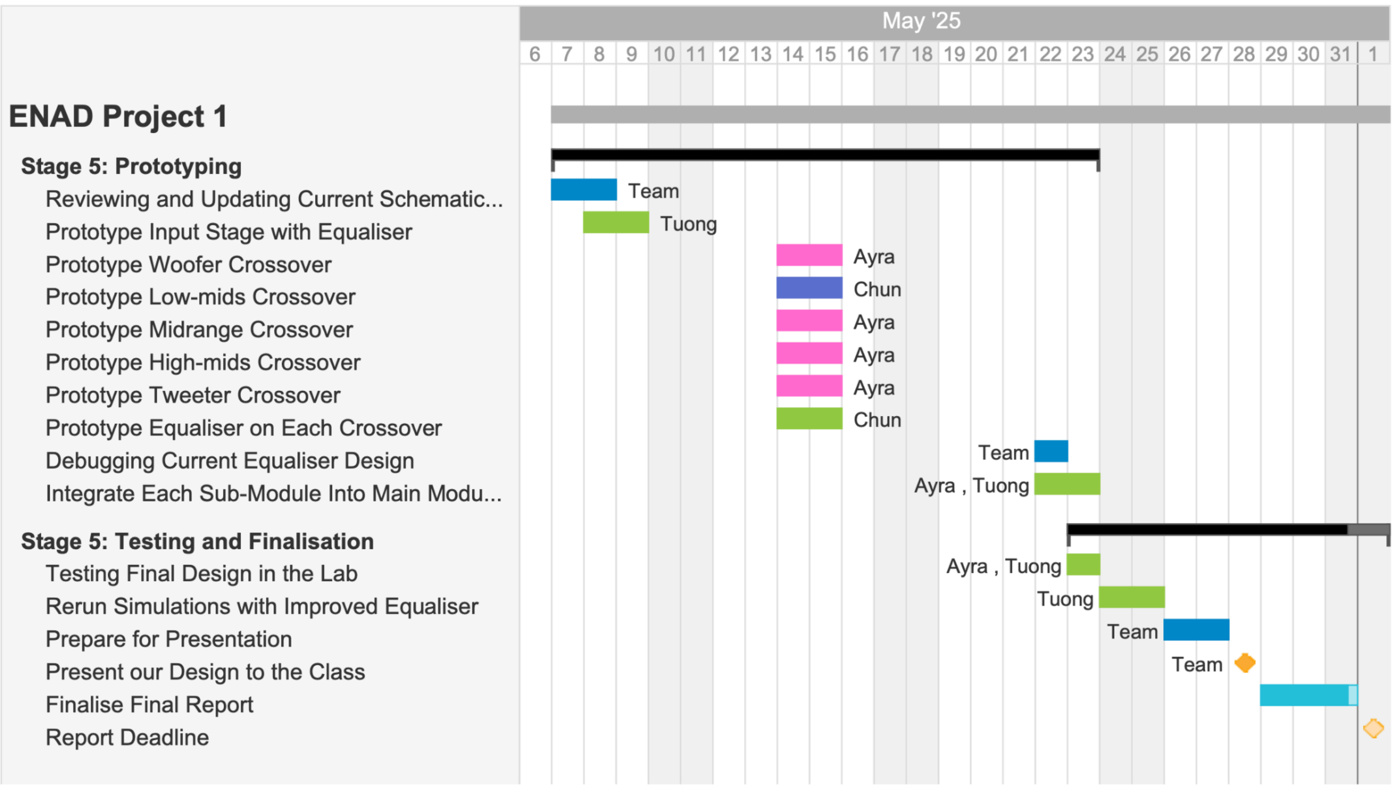
Custom Speaker with Integrated Equaliser

Note: the ‘popping’ noise heard in the video is from a group nearby.

Appendix E: Evidence of Team Collaboration & Progress Tracking

Below is a Gantt chart designed to track the progression of the project, marking notable milestones within the project timeline.





Below is a link which redirects to a word file, storing all the meeting minutes of the team.
https://1drv.ms/w/c/21bf6f25ca433f0d/EWdmUR9WSj1IoSzrA751gxsB3nlxOqZW_0ng4JOLYdRH7A